



Underwater noise due to precipitation

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PROGRAM OF

The 117th Meeting of the Acoustical Society of America

Sheraton University Inn and Syracuse University • Syracuse, New York • 22–26 May 1989

TUESDAY AM

MONDAY EVENING, 22 MAY 1989

COMSTOCK AB, 7:00 TO 9:00 P.M.

Tutorial on Acoustical Oceanography

Mauro Pierucci, Chairman

Department of Aerospace Engineering and Engineering Mechanics, San Diego State University, San Diego, California 92182

TU1. Acoustical oceanography: Child of ocean acoustics. Herman Medwin (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

Sound propagation is the preeminent technique for sensing, identifying, and communicating under the ocean surface. However, the extraordinary spatial and temporal variability of the ocean has frustrated underwater acousticians for decades. The solution, of course, was to learn more about the causes of this variability. The traditional oceanographic instruments do this rather crudely, and probably the best technique is to *use the acoustical behavior to characterize the medium*. Several years ago, ocean acoustics gave birth to acoustical oceanography. The acoustical oceanographer inverts the problem; he uses the seemingly capricious nature of sound propagation to learn about the ocean. The many successes of this young science range from the identification and counting of physical and biological inhomogeneities, such as microbubbles, zooplankton, and fish, to the remote sensing of distant rainfall and sea surface roughness, deep sea mountains, rocks and sediments, as well as the shape and strength of immense churning ocean eddies, hundreds of kilometers in extent.

TUESDAY MORNING, 23 MAY 1989

REGENCY B, 8:00 TO 11:45 A.M.

Session A. Noise I: Noise of Air-Moving Devices

Louis A. Herstein, Chairman

Tracor Applied Sciences, Inc., 7945 MacArthur Boulevard, Suite 214, Cabin John, Maryland 20818

Chairman's Introduction—8:00

Invited Papers

8:05

A1. Fan noise control—Challenges and opportunities. Hsien-sheng (Jason) Pei (Digital Equipment Corporation, 30 Forbes Road, Northboro, MA 01532)

Fans used in the cooling of mechanical and electrical equipment are often the major sources of noise. This paper reviews the past and current noise control activities in the fan industry and related trade and professional organizations. Technology trends are discussed. Research needs and priorities are identified. The impact of fan

design (blade, housing, and motor) and fan manufacturing (materials and processes) on fan noise and system acoustical quality are discussed. This paper illustrates with examples that fan noise control is more a challenge for design, application, and manufacturing than an investigation of fundamental aeroacoustic mechanisms. The approach is multi-disciplinary and comprehensive in its treatment of each integration level—from bearings and windings to motor, housing, and end-use equipment, thereby integrating noise control into fan design and manufacturing to achieve total quality for tomorrow's market.

8:30

A2. A review of air-moving device noise: Mechanisms and measurement methods. David M. Yeager (IBM Acoustics Laboratory, P. O. Box 1328, Zip 1103, Boca Raton, FL 33432)

The predominant acoustic noise sources of electronic systems are very often due to convection cooling devices. A wide range of capacities of air-moving devices (AMDs) are used in computers and business equipment, typically varying in total flow rate from 0.01 to 1.0 m³/s, and in static pressure drop from 10 to 1000 Pa. This review paper will cover several aspects of air-moving device noise: (1) basic mechanisms of aerodynamic noise generation, especially in the context of air-moving device noise, (2) measurement methods of characterizing aeroacoustic noise sources on airfoils immersed in turbulence, (3) important parameters in assessing aeroacoustic performance, (4) standardized measurement methods for determining noise emission levels of AMDs; (5) use of sound power levels of AMDs for predicting overall system noise levels; and (6) nonstandardized measurement methods for quantifying air-moving devices based on sound intensity techniques.

8:55

A3. On precise in-duct measurement of fan noise. William B. Swim (Department of Mechanical Engineering, Tennessee Technological University, Cookeville, TN 38505)

Measurement of fan noise requires extremely careful control of a multiple of variables—including fan operating point and test system configuration—to obtain useful results. Frequently, fan noise measurements are contaminated by unrecognized fan or test system changes and spurious noise sources. A new fan test system was designed to eliminate extraneous noise sources and to provide good control of the fan and the test system. This system uses round sheet metal ducts to connect the test fan to inlet and discharge anechoic terminations. Rotating microphone probes, one in the inlet and one in the discharge, are used to measure induct SPL while the fan's air performance is being determined. Two test systems, one with a mean diameters of 8 in. and the other of 16 in., have been built. Measurements have been made on both axial and centrifugal fans of sizes from 6 to 20 in. Representative results of these measurements and data from recent studies of microphone probe design and surface microphone systems use for in-duct noise measurements will be discussed. Data will also be presented on the influence of tip clearance on axial fan noise.

9:20

A4. Air-moving devices and active noise control. D. A. Olson, S. S. Wise (Digisonix Division, Nelson Industries, Inc., Stoughton, WI 53589-0200), L. J. Eriksson, and M. C. Allie (Corporate Research Department, Nelson Industries, Inc., Stoughton, WI 53589-0600)

Active noise control presents unique advantages for the reduction of noise from air-moving devices. These include excellent low-frequency performance, minimal flow restriction, and ease of installation. In this paper emphasis will be placed on the use of active noise control for silencing discharge noise from fans. A recently developed digital system using adaptive signal processing will be described. Results will be presented from a number of centrifugal and vaneaxial fans. Application guidelines have been developed to ensure that the system performance is maximized. These guidelines will be presented using results from actual field testing.

9:45-10:00
Break

10:00

A5. Sound sources of low-speed jet impingement on a large flat plate. Allan Powell (Department of Mechanical Engineering, University of Houston, Houston, TX 77204-4792)

Jet impingement on a large flat normal plate produces a noisy turbulent divergent wall jet. *As in the more general case*, the image flow in the plate can be postulated and the "three sound-pressures theorem" applied (neglecting dissipation effects): For long wavelengths ("compact" flow), the aerodynamic sound sources must be reducible to *lateral quadrupoles alone*, with maxima normal to and in that plane. Additional (usually weak ??) shear dipoles associated with the viscous traction forces lie in the plane of the plate. Specifically, the large scale axisymmetric unsteady flow associated with the flow resonances that sometimes occur resembles a periodic series of ring vortices that approach the plate and then spread out, symmetrically. Regardless of the details of this symmetrical unsteady flow, the associated source field must be periodic annular distributions of the lateral quadrupoles and traction dipoles; in the long-wavelength (compact) low-speed case these degenerate into symmetrical point quadrupoles, one lateral and one longitudinal, yielding " U^* laws" for simple similarity. Dissipation results in " U^* -type" monopoles due to heat addition fluctuations.

10:15

A6. Sound sources of high-speed jet impingement. Alan Powell (Department of Mechanical Engineering, University of Houston, Houston, TX 77204-4792)

For high-speed flows the source region is often *not compact* and the low-speed arguments are invalid: Then the motion of the flow boundary may provide insight as to the nature of the sound sources. (i) When a "two-dimensional" jet impinges on a large flat plate, the sinuous jet puffs from side to side at the plate, approximated by simple sources separated by the high impedance jet. (ii) The *high-speed* edge tone is similar. (iii) For small plates sound will be generated by the axially symmetric modulation of the strength of the unstable stand-off shock wave. Then, (a) changes in flow yield the total annular simple source strength or (b) the nearly conical boundary of the flow enveloping the plate appears to vibrate sympathetically providing an alternative source model. Some results of an alternate approach are given, in which the boundary is considered to be generating the evanescent waves, partially scattered at impingement or radiating at initiation.

10:30

A7. Suppression of shock-associated noise from improperly expanded jet flows. Darshan S. Dosanjh (Mechanical and Aerospace Engineering Department, Syracuse University, Syracuse, NY 13244)

Experimental aeroacoustic investigations of the reduction of intense noise radiated by improperly expanded turbulent jet flows are reported. The noise intensity varies with the strength of the repetitive shock structure inherently present in improperly expanded jet flows. To reduce shock-associated noise levels, the needed weakening of the repetitive shock structure is achieved when improperly expanded jet flows issue (1) from a plug nozzle with either a contoured plug or a conical perforated plug operated at a range of above-critical pressure ratios; (2) from coaxial nozzles of different configurations operated in the inverted pressure mode. Farfield acoustic results and the supporting optical records of these improperly expanded jet flows are presented. Striking changes in sound pressure level spectra, overall sound pressure level directivity, and substantial reductions in the intensity of shock-associated noise are reported.

Suppression of shock-associated noise from improperly expanded turbulent jet flows issuing from a plug nozzle operated at off-design pressure ratios is shown to be more effective than that for jet flows from an equivalent contoured convergent-divergent nozzle operated in the over- and underexpanded modes.

10:45

A8. Excitation of finite and infinite length cylindrical shells by internal sound fields. William C. Ward (Noise Control Laboratory, The Pennsylvania State University, University Park, PA 16802)

Cylindrical shells, such as industrial piping system components, are efficiently excited by broadband internal noise at discrete frequencies below the ring frequency. Two types of excitation are recognized: finite length pipe resonances that are visible in short pipes, and coincident excitation that has been studied for long or anechoically terminated pipes. Both mechanisms occur to varying degrees in pipes of any length, and both require that the acoustic and structural wavenumbers be closely (or exactly) matched. Because ducts possess an infinite number of potential coincidence frequencies, coincidence transmission (i.e., precisely matched wavenumbers) is dominant in pipes. In the limit for long shells, both mechanisms are damping controlled and approach the same levels. Experimental data for short and intermediate shells show that coincidence transmission ("infinite shell") can be the sole cause of vibration over wide frequency bands where the density of resonant modes is low. A simple theory is used to predict the frequency and amplitude of response peaks caused by both mechanisms. [Work supported by NSF.]

11:00

A9. Modeling of fluid-injection effects on flow noise. George H. Christoph (Martin Marietta Laboratories, 1450 S. Rolling Road, Baltimore, MD 21227)

Based on Naval Underwater Systems Center [communications with E. Payne of NUSC] and Soviet [Sov. Phys. Acoust. **30**(5), 394-397 (1984)] water-tunnel data, the Corcos wall-pressure model was modified to include fluid-injection effects. Boundary-layer parameters were used to model the wall-pressure frequency spectral density and the longitudinal cross-spectral density. Using predictions from a Martin Marietta boundary-layer code, favorable comparisons were made to the NUSC measured boundary-layer profiles and wall-pressure frequency spectra. Excellent agreement was obtained for both the effect of injection rate and the reduced effectiveness of fluid injection with distance from the injection source. Calculated wavenumber spectra showed that at low frequencies (less than about 200 Hz) spectral levels increased with injection, while at higher frequencies spectral levels decreased. The predicted convective peaks shifted to higher wavenumbers due to the decrease in convective velocity with increasing injection rate. The rms wall pressures are shown to increase for all fluid-injection rates. The model predicts that the large flow-noise reductions (over 20 dB) obtained in water-tunnel experiments will not be achieved in full-scale applications.

11:15

A10. Modeling of pressure-gradient effects on flow noise. George H. Christoph (Martin Marietta Laboratories, 1450 S. Rolling Road, Baltimore, MD 21227)

A methodology to predict the effects of pressure gradients on flow noise is presented. The method was developed from the pressure-gradient

data of Schloemer [J. Acoust. Soc. Am. **42**, 93-113 (1966)] and of Burton [MIT Acoustics and Vibration Lab. Rep. 70208-9 (1973)] and implemented with the Corcos wall-pressure model. Predicted boundary-layer parameters yield a wall-pressure frequency spectrum for a particular flow condition, thus allowing one to predict conditions where data do not exist. For increasing pressure flows, rms wall pressures were predicted to be unaffected for Burton's test cases but increased for Schloemer's experiments. This apparent contradiction is in agreement with the experiments. Also, it is shown that erroneous conclusions result from a zero-pressure-gradient wall-pressure model. Computations of wavenumber spectra in adverse gradients gave higher spectral levels except in the vicinity of convective peaks where levels are noticeably lower, especially at the high frequencies. For Burton's test cases, frequency-spectral-density calculations showed increases at low frequencies and reductions at high frequencies, although not as dramatic as Burton's data indicated for the most severe adverse-pressure gradients. Wall-pressure levels were predicted to increase at all frequencies for Schloemer's data. The model clearly shows

that these differences are due to the way that the adverse-pressure gradients are formed.

11:30

A11. Correlating fan performance parameters and noise. Egons K. Dunens (Control Data Corporation ARH268, 4201 N. Lexington Avenue, Arden Hills, MN 55126)

Computer cabinet designers choose fans based on their air flow characteristics but noise concerns often have a significant impact on the final cabinet design. Many physical and operating parameters can impact the fan noise. The noise emissions from a large number of fans have been measured using ANSI S12.11-1987. Regression analysis is used to evaluate the correlation of the various parameters to the fan noise level in order to understand trade-offs at the initial design stage.

TUESDAY MORNING, 23 MAY 1989

COMSTOCK B, 8:00 A.M. TO 12:00 NOON

Session B. Physical Acoustics I: Nonlinear Acoustics

David T. Blackstock, Chairman

Applied Research Laboratories, University of Texas, Austin, Texas 78713-8029

Invited Paper

8:00

B1. Nonlinear behavior leading to real rarefaction shocks and related phenomena. P. A. Thompson (Rensselaer Polytechnic Institute, Troy, NY 12181)

Early calculations by Bethe, Zel'dovich, and others have suggested the possibility of rarefaction shocks in fluids and solids. Recent experiments have demonstrated the existence of rarefaction shocks in rubber (Kolsky and Rader), quartz (Barker and Hollenbach), iron (Erkman, Ivanov, and Novikov), fluids at near-critical states (Borisov *et al.*), and in evaporating mixtures (Chaves, Thompson *et al.*). Analogous discontinuities are found in second sound in superfluid helium. The assertion that all rarefaction shocks violate the second law is untrue, both in theory and practice! Physical experiments in *fluids* leading to a single-phase rarefaction shock, or to a complete rarefaction-evaporation shock have, until now not been realized. Key ingredients for *real* rarefaction shocks include a negative nonlinearity parameter Γ , usually some degree of metastability, Chapman-Jouguet conditions, and in the case of fluids, a large molar heat capacity (many molecular degrees of freedom). Examples and related phenomena are discussed.

Contributed Papers

8:30

B2. Nonlinear wavetrains in dense gases with large specific heats. M. S. Cramer (Department of Engineering Science and Mechanics, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061), W. Pelz (Department of Mathematical Science, University of Akron, Akron, OH 44325), and L. T. Watson (Department of Computer Science, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

Periodic wavetrains propagating in fluids whose specific heats are large compared to the molar gas constant are examined. In the dense gas regime, the fundamental nonlinearity parameter $1 + B/2A$ of these fluids may become negative. The present study examines the nonclassical prop-

erties of the evolution of a sinusoidal wavetrain including the formation and propagation of expansion shocks and sonic shocks. Analytical solutions are presented for the inviscid problem and are compared to numerical solutions describing the dissipative evolution.

8:45

B3. Nonlinear propagation of sound beams in various media. E. A. Zabolotskaya (General Physics Institute of the USSR Academy of Sciences, 117942 Moscow, USSR)

Approximate nonlinear equations describing the propagation of finite amplitude sound beams in liquids, isotropic solids, and crystals are sug-

gested. The equations have been derived under the assumptions of small perturbation and slow variation of the wave shape, which are due to diffraction and nonlinearity. Diffraction is taken into account within the quasioptical approximation. Nonlinearity is considered up to quadratic terms for the cases of wave propagation in liquids and crystals, as well as for longitudinal wave propagation in solids. As for nonlinear propagation of transverse waves in an isotropic solid, cubic terms in the magnitude of the perturbation are retained.

9:00

B4. Finite amplitude effects on the propagation of a pulsed sound beam in a dissipative fluid. Kjell-Eivind Frøysa (Department of Mathematics, The University of Bergen, 5007 Bergen, Norway), Jacqueline Naze Tjøtta, and Sigve Tjøtta (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029 and Department of Mathematics, The University of Bergen, 5007 Bergen, Norway)

The linear and nonlinear propagation of a pulsed sound beam generated by a real source in a fluid is considered. The source can be plane or weakly focusing. The investigation is based on a linear and quasilinear solution of the Khokhlov-Zabolotskaya-Kuznetsov nonlinear parabolic equation. Analytical and numerical results are presented. The evolution of the pulse as it propagates from the source into the farfield region is investigated for various pulse forms. The special case of a source with distribution $\exp(-x^2/a^2)r(t)$ (x radial distance, t time) is investigated in detail, with emphasis on the role of diffraction and absorption on the self-demodulation of the pulse. The results are related to the problem of scattering of sound by sound. [Work supported by the IR&D program of ARL:UT and VISTA/STATOIL, Norway.]

9:15

B5. Effects of absorption on the scattering of sound by sound. Corinne M. Darvennes, Mark F. Hamilton (Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1063), Jacqueline Naze Tjøtta, and Sigve Tjøtta (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029, and Department of Mathematics, The University of Bergen, 5007 Bergen, Norway)

The scattering of sound by sound in a lossless fluid was discussed at an earlier meeting [Berntsen *et al.*, J. Acoust. Soc. Am. Suppl. 1 **83**, S4 (1988), and Darvennes and Hamilton, J. Acoust. Soc. Am. Suppl. 1 **83**, S4 (1988)]. Here, the effects of absorption are included. The Khokhlov-Zabolotskaya-Kuznetsov equation is used to derive farfield asymptotic results for the sum and difference frequency sound due to the noncollinear interaction of real sound beams radiated from displaced sources. There are two main contributions to the nonlinearly generated sound in the farfield: the continuously pumped sound and the scattered sound. Weak absorption affects neither the locations nor the relative amplitudes of the pumped and scattered difference frequency sound. Strong absorption attenuates the pumped difference frequency sound faster than the scattered difference frequency sound. The scattered sum frequency sound is always attenuated faster than the pumped sum frequency sound, and there may be shifts in the locations of the maxima. Numerical results are presented for the case of Gaussian primary beams. [Work supported by ONR (CMD and MFH), IR&D Program of ARL:UT, and VISTA/STATOIL (JNT and ST).]

9:30

B6. Measurement of bubble properties using a multi-frequency sound field. Anthony A. Atchley, Robert A. Perron, and Ernest R. Lineberger (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

A problem of continuous interest in underwater sound propagation is the prediction of the scattering properties of bubbles. Two important parameters in this problem are the bubble's radius and damping coefficient. A method of measuring these parameters, which is a modification of a bubble-sizing technique developed by V. L. Newhouse and P. M. Shankar [J. Acoust. Soc. Am. **75**, 1473-1477 (1984)] is described. The method exploits the nonlinear, mixing property of resonant bubbles simultaneously exposed to sound fields of different frequencies. The results of measurements of bubble radius and resonance damping coefficient are presented for bubbles having resonance frequencies less than approximately 100 kHz. Extension of the technique to smaller bubbles is discussed. [Work conducted for the Naval Coastal Systems Center and funded by the Naval Postgraduate School.]

9:45

B7. General linearized solutions of the equations of bubble pulsations. Timothy M. Cross and Robert D. Finch (Department of Mechanical Engineering, University of Houston, Houston, TX 77204-4792)

Previous attempts to find linearized solutions of the equations for vapor bubble pulsations [e.g., see Finch and Neppiras, J. Acoust. Soc. Am. **53**, 1402-1410 (1973)] assumed that oscillations would be the result of imposed sonic excitation. It was shown by Nicholas and Finch [12th Int. Cong. Acoust., Toronto, paper 14-2 (1986)] that nonlinear solutions of the equations in fact often showed exponential expansions or collapses. In this paper it is shown that if the governing equations are linearized using a Taylor series expansion, without assuming either oscillatory or exponential behavior, then there are regimes of bubble radius in which one or the other of these two is a necessary result. The simple case of a bubble with an insoluble gas content in a nonconducting liquid is considered. There can then be shown to exist a critical radius, with a value given by $2\sigma/3\gamma P$, where σ is the surface tension, γ is the ratio of specific heats, and P is the gas pressure. Below this size the behavior is exponential, above it, oscillatory. The results can be interpreted in terms of the stability of an open-loop control system.

10:00

B8. Effects of nuclei and host fluid parameters on the threshold for cavitation produced by pulsed ultrasound. Christy K. Holland and Robert E. Apfel (Engineering and Applied Sciences, Yale University, Yale Station #2159, New Haven, CT 06520)

An experimental apparatus has been developed to determine thresholds for cavitation produced in a fluid by short tone bursts of ultrasound at 0.76, 0.99, and 2.30 MHz [A. Atchley *et al.*, Ultrasonics **26**, 280-285 (1988)]. A fluid jet was used to convect potential cavitation nuclei, such as 1- μ m polystyrene spheres, echo contrast spheres, and whole blood constituents, through the focal region of the insonifying transducer. Cavitation thresholds measured with this system in water and in a fluid with ten times the viscosity of water will be presented. Cavitation was detected by a passive acoustical technique that is sensitive to sound scattered from cavitation bubbles. Results from these experiments that permit the control of nuclei and host fluid properties will be compared to an approximate theory that predicts the onset of cavitation [C. K. Holland and R. E. Apfel, Trans. IEEE UFFC-36(2) (1989)]. [Work supported by NIH, grant number 1R01-CA-39374.]

10:15

B9. Possible observation of chaos in acoustics. Bill D. Cook (Cullen College of Engineering, University of Houston, Houston, TX 77204-4792)

Approximately 20 years ago, I observed an acoustic phenomenon that could have been chaos. Not understanding what I was observing, I found

that I could maintain experimental conditions to prevent it from occurring. I did so in order to complete the task at hand. The study involved understanding what conditions subharmonics were generated in an ultrasonic standing wave cavity. Two air-backed quartz transducers were placed in a Fabry-Perot configuration with water; one transducer having a resonant frequency of about 1 MHz; the other always being at lower frequency. Their separation was adjustable to be between 5 to 10 cm apart. The 1-MHz transducer was driven with a variable frequency oscillator. At sufficient power and at some driving frequencies, subharmonics were observed both optically and with a piezoelectric transducer. Upon the onset of subharmonics, an optical diffraction pattern displayed lines in between the lines normally observed. A frequency analyzer connected to the piezoelectric transducer showed subharmonics and the corresponding idlers. Under some higher power conditions, the optical pattern became a smear and the frequency analyzer display became uninterpretable. In retrospect, I now believe what I was observing was chaos.

10:30

B10. Nonlinear aspects of the acoustic levitation of compressible spheres. Charles C. Church (National Center for Physical Acoustics, P.O. Box 847, University, MS 38677)

The use of acoustic levitation cells has become increasingly popular in both basic research and industrial materials processing. Most theoretical studies of the forces involved have assumed a linear response of the levitated object to the acoustic field. In this study some of the effects of the nonlinear responses of highly compressible spheres (i.e., gas bubbles in a liquid) on the time-average radiation force are described. The results show that, for example, as the pressure amplitude increases (1) the radius at which the force changes sign decreases; (2) the positions in the field at which the force is maximal shift relative to the positions of the maxima determined using linear theory for radii less than the fundamental resonance; (3) at a fixed position in the field, harmonic resonance responses produce local maxima in the acoustic force. The presence of the higher-order harmonic resonances also may be seen in calculations of the levitation number for a suspended bubble. Implications for levitated liquid drops will be discussed. [Work supported by NIH.]

10:45

B11. The use of acoustical levitation to study the coalescence of oil drops in water. Edward Gardner and Robert Apfel (Department of Mechanical Engineering, Yale University, 2159 Yale Station, New Haven, CT 06520)

The behavior of emulsions composed of immiscible liquids has been the object of many theoretical and experimental studies [for review, see Chen *et al.*, *AIChE J.* **30**, 622-630 (1984)]. The coalescence of a drop separated by a film from its homophase has been studied to explain the stability of emulsions; there have not been experiments, however, on free drops contacting other free drops over long periods of time due to the difficulty encountered in containing the drops. Acoustical radiation pressure provides a means of controlling free fluid drops and was considered for application to the study of coalescence. Hexane drops were trapped by a 45-kHz standing wave in a resonant water column. The time two drops remained in contact before coalescing was measured for both pure and surfactant stabilized systems. This time was then compared to theoretical predictions and nonacoustical coalescence experiments. [Work supported by the Heyl Foundation.]

11:00

B12. Finite amplitude acoustic propagation in a periodic structure. Charles E. Bradley (Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

Linear wave propagation in periodically inhomogeneous media is characterized by the division of the frequency spectrum into regions known as passbands and stop bands, the waves (called Bloch waves) associated with which are propagated and attenuated, respectively. A dispersion relation is derived for zeroth-order propagation in a rectangular waveguide, which is periodically loaded with rigidly terminated side branches. This dispersion relation exhibits both the characteristic band structure and, in the low-frequency limit, Korteweg-DeVries dispersion. For the case of finite amplitude pure tone excitation, a quasilinear analysis shows that parametric upconversion is effectively blocked regardless of whether the second harmonic frequency resides in a passband or a stop band, though the blocking mechanisms are fundamentally different. A 25.4-mm \times 38.1-mm \times 6-m waveguide was built with 38.1-mm-deep side-branches at 0.1 m intervals. Early measurements show dispersion, band structure, and second harmonic behavior qualitatively similar to analytic and numerical results. Possible applications in the study of traveling wave amplification and cnoidal wave/soliton propagation are investigated. [Work supported by Office of Naval Research.]

11:15

B13. Oscillatory motion near a shape edge. Charles Thompson (Department of Electrical Engineering, Laboratory for Advanced Computation, University of Lowell, 1 University Avenue, Lowell, MA 01854)

The problem of linear transmission of sound over a sharp edge is one of the oldest in acoustics. However, only recently has interest been taken in the nonlinear attenuation properties of such structures. A theoretical model for the scattering and dissipation acoustics will be presented. In particular, the role that vorticity generation and transport plays will be discussed.

11:30

B14. Effect of an acoustic wave on mean-flow stability. Charles Thompson and Martin Manley (Department of Electrical Engineering, Laboratory for Advanced Computation, University of Lowell, 1 University Avenue, Lowell, MA 01854)

The interaction between an acoustic wave and a mean flow near the surface of a bluff body will be discussed. The base flow solution will be presented. A nonlinear analysis will be used to find the region of validity of the linear stability solution. Nonlinear effects will be shown to stabilize disturbances outside the stagnation point region.

11:45

B15. Acoustic streaming generated by surface oscillations. Charles Thompson and Vineet Mehta (Department of Electrical Engineering, Laboratory of Advanced Computation, University of Lowell, One University Avenue, Lowell, MA 01854)

The results of a numerical analysis of acoustic streaming caused by the oscillatory flow over a two-dimensional boundary will be presented. The effect of spatial and temporal irregularity will be addressed. The validity of models of the oscillatory and time-averaged boundary layer will be explored.

Session C. Structural Acoustics and Vibration I: Raymond D. Mindlin Memorial Session

Sabih I. Hayek, Chairman

*Department of Engineering Science and Mechanics, Applied Research Laboratory, Pennsylvania State University, University Park, Pennsylvania 16802***Chairman's Introduction—8:00*****Invited Papers*****8:15****C1. One-way propagation models for linearly elastic solids.** John J. McCoy (School of Engineering and Architecture, Catholic University of America, Pangborn Hall, Washington, DC 20064)

One-way propagation models for acoustic environments have proven to be efficient and robust as bases for general purpose computer programs. The availability of a distortional mode of propagation for a linearly elastic solid environment and the coupling of this mode with the dilatational mode, in the presence of heterogeneity or of lateral surfaces of medium discontinuity, introduce additional complexity that is not yet completely understood. In this talk, an alternative derivation of one-way propagation models to that of giving algorithmic prescription to a square root of a sum of a multiplicative and a differential operator is presented. Also discussed are the difficulties encountered for linearly elastic solids. A number of numerical examples are provided.

8:40**C2. Construction of a mathematical model of the nonlinear, dynamic mechanical properties of unreinforced masonry.** Hugh D. McNiven (Department of Civil Engineering Science, University of California, Berkeley, CA 94720)

This presentation reports the results of a study of the construction of a mathematical model for predicting the nonlinear in-plane behavior of clay brick masonry walls when subjected to dynamic excitations. The construction consists of two stages: first the development of the form of the model and then the establishment of the parameter functions appearing in it using experimental data and system identification. In previous work with system identification, it was learned that there are several advantages if the system to be modeled is linear. The optimization is easy and cheap because the iterations minimizing the cost function converge rapidly. Perhaps, even more important is the fact that the transfer function itself is optimized rather than the time response of the model to a particular input, as is the case with nonlinear modeling. Experiments consist of subjecting twin walls of masonry to earthquake excitations on the shaking table at the Earthquake Engineering Research Center (EERC), University of California at Berkeley. The experiments were designed so that the structure was subjected to a series of earthquake inputs of increasing intensity, but so that consecutive intensities differed by a small amount. As a result, the response to an individual excitation, even in the nonlinear material domain (following cracking), is almost linear. It was assumed that the wall would behave isotropically. The subsequent form of the model left two parameter functions to be established, one describes elastic-plastic stresses and depends on strains; the other describes viscous stresses and is a function of strain rates. Experimental data and system identification, using a particular optimization algorithm, showed that both the shear modulus of the masonry and its associated damping factor are bilinear. The experimental time histories of acceleration and displacement were compared with those predicted by the completed model. Even in the highly nonlinear range of material behavior, the two responses were unusually close.

9:05**C3. Electroelastic interactions, biasing states, and precision crystal resonators.** H. F. Tiersten (Rensselaer Polytechnic Institute, Troy, NY 12180-3590)

In the interaction of the quasistatic electric field with deformable insulators, the condition of rotational invariance causes a combination of the electric field and the deformation gradients to occur in the constitutive equations along with the finite strain. Since the resulting system is intrinsically nonlinear, the linear dynamic equations are more general than those of linear piezoelectricity when a bias is present and reduce to them only in the absence of a bias. Even in the simplest case of stress-free thermal deformation, which is just about always present, the more general equations arise when the fixed reference coordinates at the reference temperature are

employed. The advantage of the use of reference coordinates, which cannot be employed within the usual linear theory, in the accurate calculation of the temperature sensitivity of high precision contoured quartz resonators is shown. In the treatment, the equation for the perturbation in eigenfrequency of the piezoelectric solution due to a bias, which is obtained from the more general linear equations, is employed. However, both the biasing state and the vibrational solution are obtained by solving systems of unbiased linear equations. The change in frequency resulting from any bias may readily be calculated from the perturbation equation when the linear piezoelectric solution and biasing state are known. The importance of the phenomenon of energy trapping in crystal resonators is discussed and means of controlling it are noted.

9:30

C4. Temperature measurements and photographic observations of evolving adiabatic shear bands. J. Duffy (Division of Engineering, Brown University, Providence, RI 02912)

Experiments are described in which the local temperature and local strain distribution are measured during the formation of adiabatic shear bands in steels. The specimen employed is a short thin-walled tube loaded dynamically in a torsional Kolsky bar (split-Hopkinson bar). Local temperature is determined by measuring the infrared radiation emanating at 12 neighboring points on the specimen's surface, including the shear band area. Indium-antimonide elements are employed for this purpose to give the temperature history during deformation. In addition, high-speed photographs are made of a grid pattern deposited on the specimen's surface, thus providing a measure of the strain distribution at various stages during shear band formation. The results provide a picture of the developing strain localization process, of the temperature history within the forming shear band, and of the consequence loss in the load capacity. It appears that plastic deformation follows a three-stage process that begins with a homogeneous strain state followed by a generally inhomogeneous strain distribution, and finally by a narrowing of the localization into a fine shear band. Experimental results are compared with predictions of various models.

9:55

C5. Mindlin's equations of plated, crystal plates for thin-film stress prediction. P. C. Y. Lee (Department of Civil Engineering and Operations Research, Princeton University, Princeton, NJ 08544)

When an electroded, crystal plate is subjected to a steady acceleration and/or a steady, uniform temperature change, stresses are induced due to body forces exerted on the plate and electrodes, and due to the differences in thermal expansions between the plate and electrodes. Based on Mindlin's first-order equations of plated crystal plates [R. D. Mindlin, *Progress in Applied Mechanics* (Macmillan, New York, 1963), pp. 73-84, Prager Anniversary Volume], a set of six coupled equations is obtained in which the thin-film stresses induced by the body forces and thermal expansions are taken into account. A principle of virtual displacement corresponding to these equations is also obtained. In a partially electroded, circular crystal disk, the stresses due to steady acceleration are calculated, and the changes of thickness-shear frequencies due to these stresses are predicted.

10:20

C6. Theory of superpotentials in a graded-index dielectric waveguide. Raj K. Kaul, Thomas R. Hoffend, Jr., Raymond P. McArthur (State University of New York at Buffalo, Buffalo, NY 14260), and Kenneth T. Burke (Seton Hall College, South Orange, NJ 07079)

This paper deals with the theory of dielectric waveguides in a graded-index media. The theory is first developed in terms of electrical and magnetic potentials and a gauge condition which is a generalization of the Lorentz gauge used in classical electromagnetic theory for the isotropic, homogeneous medium. The existence of these potentials under the generalized Lorentz gauge in the context of a generalized Helmholtz theorem are then proven. Even though this generalized gauge uncouples the governing equations, it does not lead to partial differential equations in the *normal* form. To alleviate this problem, superpotentials of the Hertzian variety are introduced and it is shown how equations governing the vector superpotential can be reduced to a form which is amenable to standard analysis. Then several uniqueness theorems associated with this superpotential are proven. As an illustration of the use of the superpotential and its relative merits, it is applied to certain waveguide problems in an inhomogeneous medium with variable permittivity.

C7. Surface effects on impact response of lattices. James Tasi (Department of Mechanical Engineering, State University of New York, Stony Brook, NY 11794)

An analytical and computational study is given for the influence of surface conditions on the normal impact response of two identical crystal lattices. Early studies prescribed kinematic assumptions for shock conditions at the surfaces of crystal lattices. These are compared with more recent analysis that uses solutions of the dynamic equations of motion for surfaces to determine the shock condition for impact. In the absence of any surface impurities, long-range attractive forces exist and result in adhesion of the two lattices; during which, large tensile oscillations are generated prior to the advent of a compressive region of impact. The existence of a monolayer of surface impurity on each lattice results in a more traditional shock impact condition if a slightly attractive long-range van der Waals minimum is ignored. If the latter is considered, a combination of adhesion and traditional shock impact condition are found to exist. Asymptotic solutions of the nonlinear dynamic surface equations are given for impact response and related to the early kinematic studies. [Work supported by NSF.]

C8. Abstract withdrawn.

C9. Characterization of wood for violins. Daniel W. Haines (School of Engineering, Manhattan College, Riverdale, NY 10471)

The top and back of a violin are among the most important components of the instrument, both acoustically and structurally. Centuries ago luthiers identified spruce as the ideal wood of the top and maple as the nearly unanimous choice for the back. This situation holds true today. Why is this so? The answers rests in the mechanical properties of these woods, particularly the properties that characterize the behavior of the top and back in flexural vibration. Five fundamental mechanical properties that hold most of the keys to the acoustical and structural qualities of spruce and maple are (1) density, (2) Young's modulus and (3) damping of waves traveling along the grain, and (4) Young's modulus and (5) damping of waves traveling across the grain. Shear distortion also affects the behavior of the vibrating spruce top and maple back even at low frequencies well within the audio range. Wood selection criteria and the influence of mechanical properties on the quality of the instrument will be discussed.

Session D. Engineering Acoustics I and Noise II: New Developments in Sound Power Measurement**Robert Hickling, Chairman***National Center for Physical Acoustics, University, Mississippi 38677***Chairman's Introduction—8:30*****Invited Papers*****8:35**

D1. Review of ISO and ANSI standards for the determination of sound power levels of noise sources using sound intensity measurements. Malcolm J. Crocker (Department of Mechanical Engineering, Auburn University, Auburn, AL 36830)

Sound intensity measurements can now be used to determine accurately the *in situ* sound power of machinery noise sources in the presence of high levels of background noise. Meetings of working group ISO/TC43/SC1/WG25 began in 1982 and a first draft standard was produced in late 1985. Meetings of working group ANSI S12-21 began in 1983 and produced first draft of a standard in 1984. The ANSI draft standard was circulated for balloting as a standard in 1987, and the ISO third draft was circulated in 1988. Although the ISO and ANSI drafts are similar, there are some differences. The ISO draft allows precision, engineering, and survey grade determinations and requires four indicators to be measured to decide the grade achieved. The ANSI draft only allows engineering determinations of sound power. The ISO draft only allows fixed point measurements, while the ANSI draft allows either fixed point or scanning measurements of intensity on a surface enclosing the source. Last year, 17 laboratories in North America took part in a round robin using the ANSI draft.

9:00

D2. Implementation of a second generation sound power test for production testing of earthmoving equipment. Lorne W. Tweed (Engineering Department, TT-1, Caterpillar, Inc., East Peoria, IL 61630-0945)

Caterpillar has developed an automated sound power measurement system that measures the noise of construction equipment before it leaves the assembly plant. This paper describes the test system and gives the results of verification tests conducted at various manufacturing plants around the world. The new system allows Caterpillar to quickly and accurately acquire the data necessary to ensure that their products meet noise requirements.

9:25

D3. An indoor sound-power test for light vehicles. Robert Hickling and Lee N. Bolen, Jr. (National Center for Physical Acoustics, University, MS 38677)

There is a need to develop sound-power tests as standards for the noise of manufactured products, such as automobiles. For uniformity of testing, the tests have to be independent of the test location. Sound-intensity measurements can be used for this, as has been demonstrated in tests with six automobiles. The sound power of each vehicle was measured for different loads and engine speeds, using a semicircular array of sound-intensity probes rotated around the vehicle on a dynamometer roll. The tests were conducted at Clark Laboratory Services in a large work area with a hard floor and high ceiling, with partially absorbent material on the walls. There were no room resonances apparent in the A-weighted vehicle sound-power spectra. Also, theoretical estimates with point sources in a reverberant room showed that the presence of the walls and ceiling has a negligible effect on the A-weighted sound power of the vehicles. Good agreement was obtained with a standard reference source. The results with the six vehicles indicated that the peaks in narrow-band sound-power spectra can be used to identify noisy components in a vehicle power train. [Work supported by the United States Office of Naval Research.]

D4. Estimating sound power from discrete measurements of sound intensity on a hemisphere. W. Weston Meyer (Mathematics Department, General Motors Research Laboratories, Warren, MI 48090-9055)

In theory, total sound power radiated from a noise source can be determined by integrating the normal component of sound intensity over an arbitrary enclosing surface S . In practice, one must settle for approximation, taking the sound power integral to be a weighted sum of intensity measurements at certain strategically located nodes, or meshpoints, on S . What are the strategic locations, what are the weights, and what level of accuracy can be expected? The present paper attempts to answer these questions for the particular experimental setup described in a recent publication [R. Hickling and L. N. Bolen, "Indoor system for efficient measurement of the sound power of light vehicles," NCPA Rep. RH-01-88]. Here, S is a hemisphere H plus circular base B —the latter a physical boundary that is perfectly reflective. The mesh covers H but not B : it appears rectangular in cylindrical projection, with longitudes equally spaced. Strategic positioning is therefore a matter of latitude selection. Several schemes are discussed based on different notions of optimality, the means of calculating weights for each scheme are shown, and numerical evidence is compiled to indicate that levels of accuracy attainable, by applying the quadrature formulas to various test functions (chiefly monopoles). The rapid fluctuations that can occur in the sound intensity function and the reasons for this condition are also discussed.

10:15

D5. Low-frequency problems in the determination and application of equipment sound power ratings. Peter K. Baade (Noise and Vibration Control, Inc., 171 Brookside Lane, Fayetteville, NY 13066)

There is a growing concern with low-frequency noise in buildings. Sound power ratings for the sources involved, however, typically do not include any data below the 125-Hz octave band because the present series of ANSI standards (S1.30–S1.35) and their ISO counterparts for the determination of sound power, while not setting any low-frequency limits, do not provide any information on the measurement uncertainties for frequencies below the 125-Hz octave band. In reverberation room measurements, there are at least two types of low-frequency problems: (1) difficulties in obtaining a valid space average of the sound pressure and (2) effects of the acoustic impedance seen by the source on its sound power output. The published literature on the second type contains conflicting information. This issue is important not only for standardizing equipment sound power determinations, but also for the proper application of equipment sound power ratings.

Contributed Papers

10:40

D6. ANSI round robin test on determination of sound power from sound intensity. U. S. Shirahatti, Malcolm J. Crocker (Department of Mechanical Engineering, Auburn University, Auburn, AL 36830), and Richard J. Peppin (Scantek, Inc., Rockville, MD 20850)

The use of the sound intensity technique to determine the sound power output of a source has become very popular over the years. An American National Standards Institute committee, ANSI S12-21, is currently engaged in developing a suitable standard for making sound power measurements using the sound intensity technique. The draft version of the proposed standard permits the use of both fixed point measurements and also continuous hand scanning on the enclosed surface. To check the validity of the sound power measurements and to help make suitable changes to the measurement procedure, many optional data quality indicators are also included in the draft. Recently, the working group ANSI S12-21 organized a round robin test on a standard reference sound power source. Various organizations across the United States participated in the round robin test. This paper summarizes and discusses the sound power measurement results obtained on the reference sound source.

10:55

D7. Power output and efficiency of sound production by crickets. T. G. Forrest (National Center for Physical Acoustics, University, MS 38677)

Male crickets produce calling songs that function to attract mates. Sound is produced when membranes of the wings are caused to vibrate

during stridulation. Sound fields were measured for three species of crickets. Two species of mole crickets, *Scapteriscus acletus* and *s. vicinus*, call from within burrows constructed in the soil. Sound fields of the mole crickets were hemispherical and the power output averaged $4 \mu\text{W}$ ($N = 22$, range $2\text{--}22 \mu\text{W}$). Power output was dependent upon male size and moisture content of soil surrounding the burrow. Efficiency of sound production was estimated to be less than 0.2%. Sound fields of a species of tree crickets, *Oecanthus quadripunctatus*, approximated that of a doublet source. Power output ranged from $2\text{--}17 \mu\text{W}$ ($N = 6$) and efficiency was estimated at about 1%.

11:10

D8. Automated sound power measurement system for indoor testing of motor vehicle noise. U. D. Dietschi (Clark Laboratory Services, 821 East Front Street, Buchanan, MI 49107)

Indoor noise testing of motor vehicles poses advantages not offered by the conventional outdoor pass-by test method. The sound power of a motor vehicle can be determined by measuring sound intensity over a surface enclosing the vehicle. Due to its geometric simplicity, a hemisphere was chosen for this purpose. In order to facilitate efficient measurement of sound intensity on a hemisphere, large enough to enclose an automobile situation on chassis dynamometer rolls, a special structure was designed to carry the intensity measurement probes. The structure was semicircular in shape made up of two concentric sections. The inner section with the measurement probes was hinged to the outer support structure such that it could be lowered into a position providing easy access to the microphones for calibration purposes. Truss sections were used in the

construction of the microphone boom to present as small an interference to the sound field as possible. The entire structure was suspended at its apex and was driven by a stepper motor via suitable reduction gearing. Control of the stepper motor was provided by a computer programmed to complete either continuous sweeps of the hemisphere or in a discrete manner, stopping the microphone boom at equal angular increments.

11:25

D9. Application of Gerchberg's iteration algorithm to noise spectrum restoration. Mei Q. Wu and Malcolm J. Crocker (Department of Mechanical Engineering, Auburn University, Auburn, AL 36849)

In noise control engineering, there are many cases where a steady noise is overlapped by an impulse noise. Some information about the noise source can be obtained by analyzing the frequency spectrum of the steady signal. But when the impulse signal is overlapped on the steady signal, the spectrum of the combined signal is usually different from the spectrum of the original steady signal. One way to remove the effect of the impulse is to filter it off in the time domain. By doing this, some time samples of the steady signal are removed also. The spectrum of the filtered signal may still be different from the spectrum of the original steady signal. In this paper, Gerchberg's iterative algorithm was used to restore the lost time samples and hence to restore the spectrum of the steady noise. Experiments were conducted on idealized noise sources where the spectrum of the steady noise was known. From the experimental results, it is seen that the spectrum restored by Gerchberg's iterative algorithm converges toward the original spectrum. After 25 iterations, the restored spectrum is very

similar to the original spectrum. This method only applies to bandlimited steady signals.

11:40

D10. An acoustic intensity measurement simulation for evaluating sound power determination standards. Frédéric Laville, Jean-Luc Agnan, and Jean Nicolas (Groupe d'Acoustique de l'Université de Sherbrooke, Département de Génie Mécanique, Université de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada)

Work has been done for several years on the standardization of sound power determination using acoustic intensity measurements. The validation of the standards through testing is difficult because of the large number of measurement conditions to be tested. To help overcome this problem, a computer code simulating the measurement has been developed. Whereas past attempts have been limited to special cases, the developed code is intended to handle cases that integrate more of the industrial reality. The measured noise sources as well as the extraneous noise sources are modeled by any number of monopoles for which the user specifies their quantity, source strength, location, relative phase, and degree of coherence. This approach offers computational simplicity as well as the possibility to model realistic radiation patterns such as plate radiation. The proposed ISO standard (ISO/DP 9614) is evaluated for a plate radiating in the presence of background noise. Recommended corrective actions were found appropriate for farfield measurements and inappropriate for nearfield measurements.

TUESDAY MORNING, 23 MAY 1989

NEWHOUSE II, ROOM 254, 8:30 TO 11:55 A.M.

Session E. Physiological Acoustics I, Psychological Acoustics I, and Speech Communication I: Interactions between Neurophysiology and Psychoacoustics (Sponsored in part by the AFOSR)

Christopher W. Turner, Chairman

Department of Communication Sciences and Disorders, Syracuse University, 805 S. Crouse Avenue, Syracuse, New York 13244

Chairman's Introduction—8:30

Invited Papers

8:35

E1. Evolving ideas of cochlear sound analysis and stimulus representation in hearing. Julius L. Goldstein (Central Institute for the Deaf, St. Louis, MO 63110)

Classical principles of cochlear operation and function in hearing are undergoing major revision because of recent biophysical discoveries that normal cochlear sound analysis is largely determined by centrally controlled nonlinear motor responses from the outer hair cells and that neural temporal entrainment can represent monaural stimulus information. Helmholtz's psychophysically and anatomically based hypothesis of tonotopic cochlear analysis is fully supported, but revision is required of the classical model he inspired, of that analysis as a tonotopic array of linear filters passively monitored by short-memory energy detectors. Challenges to the classical model have been presented throughout the history of auditory science from psychophysical studies of combination tones, idiotones, masking, and periodicity pitch. Hence, it is proposed that psychophysics can now be exploited systematically and interactively with biophysical knowledge to contribute to developing the required revised cochlear principles. To get beyond the establishment of correlations between

psychophysical and biophysical data for specific phenomena, models of some generality are needed for cochlear nonclassical responses and for psychophysical measurement. It is proposed that ideal observer theory provides a general working hypothesis that has been successful in filling the second need. A new signal processing model for nonlinear and active cochlear frequency analysis was formulated to fill the first need. Two modeling studies of the relationship between psychophysics and physiology will be described in detail for periodicity pitch and nonlinear masking.

9:15

E2. Effects of duration on intensity discrimination: Psychophysical data and predictions from single-cell response. R. R. Fay, W. P. Shofner, and R. H. Dye (Parmlly Hearing Institute, Loyola University of Chicago, 6525 North Sheridan Road, Chicago, IL 60626)

An ROC analysis was performed on responses of single auditory-nerve fibers (goldfish) and cochlear nucleus cells (gerbil) in order to predict intensity discrimination (in the goldfish and human) as a function of signal duration. To evaluate that the mean and variability of spike counts within single units account for psychophysical performance, spike number distributions were obtained ($N = 100$) for several durations (20 to 400 ms) and level differences (0.5–4 dB) at a unit's best frequency. The percent correct performance based on spike counts was found by generating ROC curves from empirical distributions and computing the area under the ROC [$P(A)$]. Theoretical psychometric functions were compared with psychometric functions from human and goldfish listeners obtained using a 2IFC paradigm (human) and a rating method in classical respiratory conditioning (goldfish). The forms of the neural and psychophysical duration functions are similar in the mammal and the fish, but the fish shows higher thresholds compared with the human and with the neurophysiological predictions. In general, psychophysical performance is well modeled by the optimum processing of spike counts from individual cells. [Work supported by a Center Grant from NINCDS.]

9:45

E3. Physiological correlates of forward masking in single nerve-fiber and compound neural responses recorded from the auditory nerve. Evan M. Relkin, John R. Doucet, Robert L. Smith (Department of Bioengineering, Institute for Sensory Research, Syracuse University, Syracuse, NY 13244-5290), and Christopher W. Turner (Department of Communication Sciences and Disorders, Institute for Sensory Research, Syracuse, NY 13244-5290)

Masking can be defined as a reduction in the detectability of one sound stimulus, the probe, due to the presence of a second sound stimulus, the masker. Reduction in detectability can be quantified by measuring the increase in threshold for the detection of the probe tone. Most previous studies of the correlates of forward masking in the auditory nerve have measured reductions in averaged responses to the probe produced by the masker. It is not possible to relate reductions in averaged responses to changes in detectability since detectability depends on response magnitude and variance. However, two-interval forced procedures can be used to measure detection thresholds for spike trains of primary, single nerve fibers [Relkin and Pelli, *J. Acoust. Soc. Am.* **82**, 1679–1691 (1987)] and N_1 of the compound action potential (CAP) [Relkin and Smith, *J. Acoust. Soc. Am. Suppl.* **1** **83**, S98 (1988)]. Relkin and Turner [*J. Acoust. Soc. Am.* **84**, 584–591 (1988)] showed that there are large discrepancies between behavioral thresholds and thresholds for single nerve fibers for forward masking. Particularly at high masker levels, thresholds for the detection of the probe in the activity of single fibers are more sensitive than those measured behaviorally. Several hypotheses that might explain this discrepancy include the effects of spatial and/or temporal processing of the responses of single neurons. These hypotheses are beginning to be tested using both single fiber and compound neural recordings. Forward masking of the CAP is more similar to behavioral forward masking than is forward masking in single neurons, suggesting a spatial summation effect. Methods for measuring peristimulus compound potentials have also been developed to investigate temporal effects such as the relative importance of onset and steady-state responses.

10:15–10:25

Break

E4. Physiological mechanisms of masking and intensity discrimination. Bertrand Delgutte (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139, and Eaton-Peabody Laboratory of Auditory Physiology, Massachusetts Eye and Ear Infirmary, Boston, MA 02114)

Recent electrophysiological studies in which stimulus paradigms were designed to mimic those of psychophysics suggest that a broad range of psychophysical data on masking and intensity discrimination can be simply explained by assuming that signal detection is primarily based on average discharge rate information in auditory-nerve fibers, provided that information from high-threshold fibers is taken into account. These studies further suggest that tone-on-tone masking is due both to spread of the excitation produced by the masker to the signal place along the cochlea, and to suppression of the signal by the masker, with the relative importance of the two mechanisms depending on masker level and frequency separation between signal and masker. In particular, two-tone rate suppression is largely responsible for the upward spread of masking at high masker levels and for differences between simultaneous and nonsimultaneous masking techniques. New results will also be presented on unmasking due to suppression of the masker by the signal, improvements in signal detectability by off-frequency listening, and the role of olivocochlear efferents in signal detection. [Work supported by NIH Grant NS13126.]

E5. Interactions between neurophysiology and speech discrimination. Donal G. Sinex, Lynn P. McDonald, and John B. Mott (University of California, Los Alamos National Laboratory, Physiology Group, M882, Los Alamos, NM 87545)

Psychophysical approaches have provided much information about the processing of speech sounds by humans and also by laboratory animals. These studies establish the essential functions of the auditory system that must be explained and define the limits over which they operate. However, processing mechanisms cannot be discovered by psychophysical approaches. Details about mechanisms must come from neurophysiological studies. This view has guided this investigation of the processing of voice onset time (VOT), a complex acoustic cue that differentiates between consonants such as /d/ and /t/. The peripheral neural representation of VOT syllables was studied, asking which (if any) properties of the neural responses correlated with the results of psychophysical experiments conducted by Kuhl and Miller [J. Acoust. Soc. Am. **63**, 905-917 (1978); **70**, 340-349 (1981)]. The psychophysical findings provided a standard against which hypotheses about the representation of VOT information could be evaluated. As a result, it was possible to identify a neural mechanism in the auditory periphery that may contribute to the formation of phonetic categories based on voicing. [Work supported by NS23242 from NINCDS.]

E6. Problems and opportunities in extending psychophysical/physiological correlation into the central nervous system. Eric D. Young (Department of Biomedical Engineering and Center for Hearing Sciences, The Johns Hopkins University, Baltimore, MD 21205)

Because auditory-nerve fibers form a relatively homogeneous population, it is possible to build models that account for psychophysical performance on the basis of auditory-nerve response properties using rather direct assumptions. However, when this analysis is extended to the CNS, the problems become more complex because of the diversity of different response types and the multiplicity of parallel central pathways. At minimum, one is faced with parallel analyses of several systems, some of which may not be relevant to the psychophysical problem being analyzed. One approach has been to use natural stimuli whose behavioral significance for the animal is clear and to analyze behavioral and physiological responses to those stimuli in the same animal. Examples of successful application of this approach are bat sonar and barn-owl sound localization. Another approach is to use the properties of directionally selective spectral shaping by the pinna to define stimuli with known behavioral relevance. Examples taken from the cat cochlear nucleus will be used to illustrate the issues raised above. [Work supported by Grant NS12524 from NINCDS/NIDCD.]

Session F. Architectural Acoustics I: Room Acoustics for Film and Television Production, Post-Production, and Cinema Spaces

Elizabeth A. Cohen, Chairman

Charles M. Salter Associates, Inc., 930 Montgomery Street, San Francisco, California 94133

Chairman's Introduction—8:45

Invited Papers

8:50

F1. Room acoustics for THX[®] motion-picture theaters. Tomlinson Holman (School of Cinema-Television, University of Southern California, Los Angeles, CA 90089-2211 and Lucasfilm Ltd., Box 2009, San Rafael, CA 94912)

The THX[®] Sound System is installed in over 300 premium motion-picture theaters and dubbing facilities worldwide. An important part of the program is specified room acoustics, including both local acoustical environment of loudspeakers and global room acoustics issues including background and intrusive noise control and reverberation time criteria. Acoustical standards established for this program will be examined, and the methods of measurement will be explained. Time-honored techniques such as using appropriate measurement microphones for the sound fields encountered, as well as time and space averaging, are applied in new, more portable, forms.

9:30

F2. Temporal versus spatial considerations in reference loudspeaker systems. George L. Augspurger (Perception, Incorporated, Box 39536, Los Angeles, CA 90039)

Until recently, the performance of loudspeaker systems used for music mixdown or cinema reproduction has been specified in terms of on-axis anechoic frequency response or 1/3-oct room response. Now, readily available test methods allow measurement of delay characteristics as easily as conventional frequency response. So-called "time-corrected" playback systems, in fact, exhibit a wide range of temporal and spatial behavior. Theoretical and measured data are presented that indicate strong interaction between spatial side effects and listening room acoustics. Subjective and objective trade-offs are discussed for three typical playback situations.

10:10

F3. Electroacoustics in the OMNIMAX[®] theater—Challenge and opportunity. Alvis F. Wales (Sonics Associates, Inc., 237 Oxmoor Circle, Birmingham, AL 35209)

IMAX[®] and OMNIMAX[®] films have significantly advanced the realism and impact of motion picture presentations. To derive full benefit from the giant screen, the theater acoustics and sound reproduction system must be equal in quality to the picture. Most IMAX and OMNIMAX theaters have seen new construction, which permits a higher level of acoustical performance and noise control than most other cinemas. OMNIMAX theaters, in particular, pose special acoustical problems because of the focusing effect of the dome screen, combined with the tendency of architects to choose an acoustically difficult shape (such as a cylinder, a hemisphere, or even a sphere) for the enclosing building. To permit full spatial and directional effects, the theater acoustics are designed to be relatively nonreverberant, leaving the "ambience" and sound "images" to be created by the sound track rather than dominated by the theater acoustics. Particular attention is paid to the sound system in terms of frequency response, dynamic range, and imaging capability.

10:50

F4. The reflection phase grating diffusor: A five-year progress report. Peter D'Antonio (RPG Diffusor Systems, Inc., 12003 Wimbledon Street, Largo, MD 20772)

The reflection phase grating diffusor, RPG, which is based on mathematical number theory sequences suggested by Manfred Schroeder, has brought to architectural acoustics a missing design ingredient—broadband wide-angle sound diffusion—and opened up many possibilities for acoustical consultants and architects. The RPG diffusor system was first introduced into the recording industry in 1983 and in the short span of 5 years has become a standard ingredient in hundreds of forefront recording/broadcast facilities, worship spaces, corporate conference/teleconference and A/V facilities, performing arts facilities, and home listening rooms. A few installations include The Oak Ridge Boys, Peter Gabriel, Whitney Houston, CBS Records, Telarc, DMP, Polygram, CBS-TV, NBC, CBC, BBC, WQXR, WFMT, Houston Grand Opera, Chrysler Hall, Kentucky Center for the Performing Arts, Word of Faith Ministries, Crenshaw Christian Church, Clear Lake United Methodist Church, Duke University, SUNY, and Lakeland College. This presentation will comprehensively review the theoretical and experimental performance characteristics of the RPG and examine a broad spectrum of applications. Many case study photos and TEF time and frequency response measurements will be presented.

11:05

F5. Assessing electroacoustic systems with auditorium acoustics measures. J. S. Bradley and R. E. Halliwell (Institute for Research in Construction, National Research Council, Montreal Road, Ottawa, Ontario K1A 0R6, Canada)

Modern auditorium acoustics measures have been successfully used to objectively assess changes to auditoria. This paper reports the use of the same techniques to assess the acoustical changes introduced in halls by three types of electroacoustic systems. These were: a film sound system and two different room acoustics modification systems. The RAMSoft room acoustics measurement program was used with broadband exponential pulses as the source signal. The program calculates 12 different quantities in each of six octave bands and permits an evaluation in terms of the needs for both speech and music. While one of the room modification systems was not found to be very effective, the other produced considerable increases in reverberation along with some related effects. Repositioning of the film sound system loudspeakers was shown to improve the performance of the system. The results clearly demonstrate the value of objective evaluations of electroacoustic systems in rooms.

11:20

F6. Electronic equipment noise in video facilities. Timothy J. Foulkes (Cavanaugh Tocci Associates, Inc., 327 F Boston Post Road, Sudbury, MA 01776)

Noise from electronic equipment is a significant problem for many video studios and ancillary facilities. Frequent efforts to control room ventilation noise are negated by noisy components that must be located within the critical spaces themselves. This paper will review acoustical design considerations for a few recent case histories. Measured data from several completed facilities and strategies for noise control will be discussed.

11:35–12:00

Panel Discussion

TUESDAY MORNING, 23 MAY 1989

REGENCY C, 9:00 TO 11:35 A.M.

Session G. Underwater Acoustics I: Acoustic Fields

Stanley M. Flatté, Chairman

Physics Department, University of California, Santa Cruz, California 95064

Chairman's Introduction—9:00

Contributed Papers

9:05

G1. Matched field processing for a moving source. John M. Ozard, Gary H. Brooke, and Scott Tinis (Defence Research Establishment Pacific, FMO Victoria, British Columbia V0S 1B0, Canada)

The detection or localization of a sound source in a multipath or multimode environment is improved by taking the multipaths into account

through matched field processing (MFP). Frequency domain MFP (FDMFP) is very effective for a stationary source. However, there is an implicit averaging over time in FDMFP through the use of the Fourier transform and a further averaging if the covariance matrix is formed. Consequently, if the sound source is moving, the averaging of the time dependent signal will lead to an increase in ambiguity of the source position and a loss in array gain. When the source is moving, the received

signal contains information about both source position and source motion. To take full advantage of the information in the time sequence, MFP in the time domain (TDMFP) was implemented. TDMFP is equivalent to obtaining the narrow-band gain and MFP gain in one step through the use of a Fourier transform modified by the propagation. Simulated results confirm the improved localization and array gain of TDMFP compared to FDMFP for a moving source.

9:20

G2. A stable data adaptive method for matched-field array processing in acoustic waveguides. C. L. Byrne, R. I. Brent (Department of Mathematics, University of Lowell, Lowell, MA 01854), C. Feuillade (SYNTEK, Inc., 2101 E. Jefferson Street, Rockville, MD 20852), and D. R. DelBalzo (Naval Ocean Research and Development Activity, Stennis Space Center, MS 39529-5004)

The presence of a "modal noise" component leads to estimator instability when Capon's maximum likelihood (ML) method is applied to the processing of data from a vertical array in an acoustic waveguide. The physics of the waveguide forces signal vectors and noise vectors alike to be projected onto the span of the "mode" vectors, when the number of sensors (N) exceeds the number of propagating modes (M). The instability occurs whenever the (single snapshot) $N \times 1$ data vectors have the form $\mathbf{x} = U\mathbf{s} + U\boldsymbol{\gamma} + \text{white noise}$, where the matrix U is $N \times M$ (sampling the normal modes at the hydrophone locations and independent of the actual acoustic disturbances present), and \mathbf{s} and $\boldsymbol{\gamma}$ correspond to signal and ambient noise sources, respectively. This condition arises in normal-mode and local normal-mode propagation. The dominant eigenvectors of R^{-1} (where R is the cross-spectral matrix) are sensitive to slight inaccuracies in the calculation of R^{-1} in ways that affect the performance of the ML estimator. Following transformation of the $N \times N$ matrix R to the $M \times M$ modal space cross-spectral matrix T , Capon's method is applied to T to obtain the "reduced maximum likelihood" (RML) estimator. This procedure, which is a development of the sector focused stability technique of Steele and Byrne [Proceed. ISSPA 87, 24-28 August 1987, Brisbane, Australia, pp. 408-412], largely eliminates instabilities due to inaccurate inversion of R . Simulations are presented for a shallow-water environment to provide comparison between the ML and the RML estimators. These indicate that the degree of instability depends upon the level of noise (both correlated noise and white noise) and that a significant improvement in performance can be expected by use of the RML estimator in both cases.

9:35

G3. A symmetry renormalization method for matched-mode sidelobe reduction. George B. Smith and George M. Frichter, IV (Naval Ocean Research and Development Activity, Stennis Space Center, MS 39529-5004)

Current matched-field research at NORDA is centered on techniques that attempt to match measured and predicted modal amplitudes for improved detection and localization of acoustic sources in shallow-water waveguides. Ambiguity functions generated by these modal estimators display a sidelobe structure that is symmetric about the true source peak. This symmetry represents additional information about the signal location, which can be used to further enhance detection. Here, a simple correlation algorithm is presented which enhances the signal peak and suppresses sidelobes by renormalizing each point of the ambiguity function in accordance with the symmetry around that point. Since a renormalized ambiguity function retains the range symmetry of the original, the technique can (within limits) be applied iteratively. Computer simulations of a shallow-water Pekeris waveguide are used to demonstrate the effectiveness of renormalization when applied to both narrow-band and frequency-averaged mode matching.

9:50

G4. Source localization: Matched field versus matched mode. Synthetic and real data performance analysis. Sergio M. Jesus and Rachel M. Hamson (SACLANT Undersea Research Centre, I-19026 La Spezia, Italy)

The present study compares the matched field and the matched mode techniques for passively localizing a narrow-band point source in a shallow-water, range-independent environment. The matched mode technique is fully characterized in terms of sidelobe ambiguity performance and robustness against both system and environment parameter variation and mismatch. Comparative results are also shown for real data detection of a cw source immersed at different depths in a 120-m depth channel using a 62-m aperture vertical array. The results of this study indicate that the matched mode method is much less sensitive to the environmental conditions than the matched field method, and in particular, the result is less degraded by the effects of partial water column sampling (short array). Results obtained on real data showed good agreement with the corresponding tests from simulated data. However, a large sidelobe coverage was found for some situations leading to detection losses. Major causes of performance degradation are the uncertainty on the array sensor position due to array motion and correlated noise due, mainly, to surface-generated noise.

10:05

G5. Broadband acoustic-field simulations from standard ray theory. Stanley M. Flatté, John Colosi, Timothy F. Duda, Galina Rovner, and Jan Martin (Physics Department, University of California, Santa Cruz, CA 95064)

The complete wave field over a small region around 1000 km from a pulsed source is reconstructed in two ways. First, all the rays from the source to a vertical array of receivers at 1000 km are found, along with their travel times, number of caustics, arrival angles, and intensities. The pattern of wave fronts in a space at a given time is then reconstructed on a closely spaced grid surrounding 1000 km by treating these rays in an appropriate way. Second, the parabolic equation method is used at multiple frequencies to synthesize a pulse. The two fields are compared. Finally, the effect of internal waves is simulated by use of the first method, introducing random fluctuations on the travel times and arrival angles of each ray. [Work supported by ONR, Code 1125OA.]

10:20

G6. Acoustic wave front distortions at long ranges from internal waves. Timothy F. Duda and Stanley M. Flatté (Physics Department, University of California, Santa Cruz, CA 95064)

In the absence of small-scale variations on sound speed in the ocean, a pulsed source delivers a series of smooth wave fronts onto a vertical array at long range from the source (multiray propagation). Small-scale variations such as internal waves induce distortions on the wave fronts with transverse correlation function determined by the phase-structure function, which is itself calculable by integrating appropriate functions along the trajectory of an undistorted ray. Expressions for the phase-structure function at small separations have been previously given in the form of arrival-angle spreads due to internal waves, but these expressions are only for vertical receiver separations up to about 100 m. Evaluations of the phase-structure function for separations up to several kilometers are presented, and particular realizations of wave front distortions that result from these internal-wave effects are shown. [Work supported by ONR, Code 1125OA.]

10:35

G7. Preliminary results from the 1988 Monterey Bay acoustic tomography experiment. James H. Miller (Department of Electrical and Computer Engineering, Code 62Mr, Naval Postgraduate School, Monterey, CA 93943), James F. Lynch (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543), and Ching-Sang Chiu (Department of Oceanography, Naval Postgraduate School, Monterey, CA 93943)

An ocean acoustic tomography experiment was held 12-16 December 1988 near Monterey, California. The objectives of this experiment were to test a tomographic system to analyze the effects of ocean surface waves, internal waves, and complex three-dimensional bathymetry on long-range acoustic propagation. An acoustic source with a center frequency of 224 Hz and source level of 177 dB was placed on a seamount 30 km off Point Sur. Seven modified sonobuoys (with anchor, bottom-mounted hydrophones, large capacity batteries, and large floats) were placed in Monterey Bay to receive the acoustic signals at ranges of 35 to 60 km from the source. The sonobuoy rf signals were received, demodulated, and the acoustic data were recorded on shore. Oceanographic measurements were taken (for comparison with the acoustically derived results) with a surface wave frequency-directional spectra buoy, surface wave frequency spectra buoys, CTD yo-yo's for internal wave spectra, ADCP, and conventional hydrographic survey for sound-speed profiles. Preliminary analyses and comparison of the acoustic and oceanographic measurements will be presented. An outline of what the complete analysis will entail will also be presented. [Work supported by ONR and Naval Postgraduate School Research Council.]

10:50

G8. Sound-speed determination using matched field processing techniques. C. Karangelen (International Business Machines Corporation, Manassas, VA 22110) and O. Diachok (Code 5120, Naval Research Laboratory, Washington, DC 20375-5000)

Matched field processing is shown to be effective for estimating sound speed in a deep, range-independent ocean environment. The amplitude and phase of signals from a distant source measured on a large aperture vertical array are sensitive to changes in sound speed. This sensitivity is exploited to infer the environmental sound-speed profile by matching predicted and measured amplitude and phase. A sound-speed profile model is developed based on a modified version of Munk's canonical sound field equation. This model is used to determine sound-speed profile using a 15-Hz signal from a 240-m explosive source detected on a 675-m vertical array at a range of 50 km in a deep-water Pacific environment, character-

ized by a classical range-independent sound channel at 700 m. The search for the best estimated profile is conducted by varying the sound channel axis strength and depth in the modified Munk equation while maintaining a constant sound-speed profile at great depths. Differences between the estimated and measured profiles are less than ± 2 m/s; the sound channel axis depth is determined within 20 m of the measured axis depth. Extension of this approach to include mesoscale features and greater range is discussed.

11:05

G9. Localization schemes for beam-type sources. I. T. Lu (Department of Electrical Engineering and Computer Science/Weber Research Institute, Polytechnic University, Farmingdale, NY 11735)

A time-harmonic isotropic source in stratified waveguides can be localized by performing matched-field processing in "mode space." Because the range and depth information of the source are contained only in the phase and magnitude, respectively, of complex model amplitudes, the range and depth data can be processed independently. This is not true for beam-type sources. When propagating in a waveguide, an initially collimated beam undergoes diffusion after successive reflections and refractions and is converted eventually into the oscillatory pattern of one or more guided modes. However, the upgoing and downgoing plane wave constituents of a given mode do not have the same excitation strength as in the case of isotropic sources. Here, localization schemes of a Gaussian beam source that is modeled via the complex source point technique are considered. The beam direction, beam width parameter, and waist location are determined in "mode space." The procedure is greatly simplified if the source is a well-collimated beam. [Work supported by NSF.]

11:20

G10. Source localization by inversion of the parabolic equation method. Susan M. Bates (MS 171, Raytheon Submarine Signal Division, 1847 West Main Road, Portsmouth, RI 02871-1087) and Bruce J. Bates (Building 1171, Naval Underwater Systems Center, Newport, RI 02841)

The inverse split-step parabolic equation method is derived. Spatial localization of single and multiple low-frequency harmonic point sources in a deep ocean is demonstrated by using simulated pressure field measurements from a vertical array. In addition, the method is applied to a source extended in range and depth. This method is solved in a single iteration, unlike some inverse techniques that vary parameters.

Session H. Engineering Acoustics II and Physical Acoustics II: Warren P. Mason Memorial Session

Harry B. Miller, Chairman

Naval Underwater Systems Center, New London, Connecticut 06320

Chairman's Introduction—1:00

Invited Papers

1:05

H1. Warren P. Mason (1900–1986), physicist, engineer, inventor, author, teacher. Robert N. Thurston, David A. Smith, Jane Baran, Kwok Cheung, and John Johnson (Bellcore, Room 3X-223, Red Bank, NJ 07701-7040)

Warren Perry Mason, a Charter Member, Fellow, President, and Gold Medalist of the Acoustical Society of America, consistently applied his understanding of fundamentals to explain physical processes and create practical devices. As a physicist, he led us to a better understanding of fundamental effects in liquids and solids. He made the first measurement of shear elasticity in liquids and helped establish the type of motion that polymer chains can make. In solids, he contributed to quantitative understandings of phonon drag on charge carriers in semiconductors, fatigue of metals, and damping of acoustic waves in metals, insulators, semiconductors, alloys, and rocks. As an engineer and inventor, he led advances in mufflers and noise control, electromechanical filters for carrier-frequency telephony, piezoelectric crystals and ceramics for electromechanical transducers, and semiconductor strain gauges. With about 200 patents, he is the most prolific inventor in the history of Bell Labs. As an author and teacher, he wrote over 200 papers and 4 reference books that teach fundamental concepts, give complete tensorial descriptions of numerous physical interactions in crystals, describe research results, and guide the reader to the related literature. This talk will sample Mason's contributions to physical acoustics and will give an example of a communications device made at Bellcore, in which Mason would surely have been interested, namely an acoustically tuned optical filter [B. L. Heffner *et al.*, *Electron. Lett.* **24**, 1562–1563 (1988)].

1:30

H2. An equivalent circuit appreciation of Warren P. Mason. Arthur Ballato (U.S. Army Electronics Technology & Devices Laboratory LABCOR, Fort Monmouth, NJ 07703-5000)

Warren Mason was a man of many parts and prolific in them all. One area in which he exercised a decisive influence was that of equivalent network representations of electromechanical systems. This paper salutes Mason's accomplishments in helping to bridge the disciplines of acoustics and electronics. It begins with a brief discussion of analogs and describes the Butterworth–VanDyke circuit of a piezoelectric vibrator. This is where Mason started his productive work in the subject, introducing acoustic transmission lines, mechanical ports, and piezoelectric transformers. Today, the Mason equivalent circuit is universally used for bulk and surface acoustic wave device characterization. It has also given rise to a variety of alternative formulations such as analog networks, the KLM equivalent circuit, and systems models, which are discussed.

1:55

H3. Piezoelectric ceramic compositional development. Don Berlincourt (Channel Products, Inc., 21 Kenton Road, Chagrin Falls, OH 44022)

The original piezoelectric ceramic material was unmodified barium titanate. It was little used except as compositionally modified. The earliest work was by W. P. Mason, and this led to improved characteristics for sonar transducers and then to phonograph cartridge applications. By the mid 1950s, lead titanate zirconate materials were shown by B. Jaffe to have higher piezoelectric coupling and application at a much higher temperature. Over the next 10 years, many modified compositions were developed. These led to much improved sonar systems and ultrasonic cleaners and to applications in ultrasonic bonders, stereo phonograph cartridges and even printers. The new compositions also made possible applications such as piezoelectric ceramic filters, gas ignition devices, and camera flashbulb actuators. More recently, specialized ceramics have been developed based on lead titanate and lead metaniobate, but major efforts have been directed to applications of lead titanate zirconate compositions, which now touch virtually every home and automobile. The history of compositional studies with ferroelectric ceramics is reviewed and the types of characteristics achieved are summarized. The compositional additives and some general principles to explain their behavior are discussed.

H4. Acoustical filters in plane-wave fields. Richard K. Cook (4111 Bel Pre Road, Rockville, MD 20853)

Lumped-parameter acoustical filters were thoroughly described by Warren P. Mason in his book on electromechanical transducers and wave filters. Today, the filtering action of open-ended circular (standing-wave) tubes and of plane-parallel waveguides is described. The description is most simply in terms of the spatial wave fields of open-ended tubes and slots. These were first analyzed accurately by L. A. Weinstein (Vaynshteyn) in the 1940s–1950s for both acoustical and electromagnetic waves. The basic mathematical technique was detailed examination of the diffracted field as a wave emerged from the open end after propagation inside the waveguide. The solutions to the diffraction problems are applied, in acoustical engineering, to the accurate analysis of open resonators, as well as to the analysis of probe tubes used for measurement of sound fields. Earlier investigators had been able to do no more than arrive at rough approximations in the form of “end corrections” to the length of a resonating circular tube, flanged at its open end.

2:45

H5. Using piezoelectric film and ultrasound resonance to determine the complete elastic tensor in one measurement. J. D. Maynard (The Pennsylvania State University, 104 Davey Lab, University Park, PA 16802)

The ultrasonic and elastic properties of materials is conventionally measured using quartz, lithium niobate, etc., transducers and a pulse-echo technique with the transducer driven at resonance. Some problems include transducer ringing, transducer bonding, parallelism of sample faces, beam diffraction, and the necessity of remounting transducers in order to measure all of the elastic constants. Usually these problems can be minimized, but, with samples that are only a fraction of a millimeter in size, conventional ultrasound measurement becomes difficult, if not impossible. However, nearly all of these problems disappear if a resonance technique is used, and all of the elastic constants may be determined with a single measurement. For the broadband response and minimum loading by the transducer required for a resonance measurement in a small sample, polyvinylidene fluoride (PVDF) piezoelectric film as thin as $9\text{ }\mu\text{m}$ is ideally suitable. Small active areas and leads are produced with metalization patterns on each side of the PVDF film. For resonance measurements, electrical cross talk across the small sample is processed by frequency modulating the drive and using phase sensitive detection. Samples with dimensions of only a few hundred microns may be measured with large signal-to-noise ratios. [Work supported by the Office of Naval Research and NSF Grant DMR 8701682.]

3:10

H6. Acoustic cavitation 42 years after the Briggs, Johnson, and Mason paper. Robert E. Apfel (Yale University, Department of Mechanical Engineering, P. O. Box 2159, New Haven, CT 06520)

Warren Mason was one of the early pioneers of high-power ultrasonics who studied its effects on liquids [Briggs *et al.*, *J. Acoust. Soc. Am.* **19**, 664–677 (1947)]. One of these effects is acoustically induced bubble activity. Such activity can be desirable, as in ultrasonic cleaning, or undesirable, as with cavitation on sonar transducers. This talk will review the basic physics underlying the onset of cavitation, the dynamics of bubble motion, and the effects caused by cavitation, with emphasis on how to optimize or minimize these effects, depending on the application. [Much of this work has been supported by the Office of Naval Research and by the National Institutes of Health through Grant 1R01CA39374.]

3:35

H7. The Mason horn, extension and applications. D. N. Beshers (Henry Krumb School of Mines, Columbia University, New York, NY 10027)

Warren P. Mason, in the mid 1950s, introduced the inverted exponential horn as a concentrator of ultrasonic waves. In 1968, he published an extended account of a stepped approximation to the exponential horn. Part of the horn was replaced by a half-wavelength piece that functioned as a mechanical transformer with a step reduction in diameter at the quarter-wave point. The specimen was shaped like a dumbbell to give a mass-spring-mass resonance with a further step down in diameter. The result was a substantial reduction in area and thus an increase in vibratory stress. The apparatus may also be described as a composite resonant oscillator: Each of the three elements, transducer, transformer, and specimen, is at resonance. Mason developed the theory of this stepped horn only for perfect resonance, and allowed for damping only in the reduced part of the specimen. Here, his theory is extended to allow for small deviations from resonance, such as must occur in practice, and for damping in the other elements. When the damping and the deviations from resonance are small, simple sum rules hold. Variations in specimen design are also considered. Applications, past and future, will be discussed.

H8. Resonance of a parallelepiped to determine single-crystal elastic constants up to 1800 K. Orson L. Anderson (Institute of Geophysics and Planetary Physics, University of California at Los Angeles, Los Angeles, CA 90024-1567)

The experimental free oscillation spectrum of a rectangular parallelepiped specimen of a single crystal has been obtained on corundum up to 1800 K, using buffer rods to separate the specimen from transducers. By inversion techniques, a theoretical spectrum can be generated, which depends upon the elastic constants, crystal symmetry, and dimensions. Using computational techniques involving the inversion of large matrices, a set of elastic constants is found that produces a theoretical spectrum matching the experimental spectrum out to about 40 modes. Using this technique, the mineral physics laboratory of UCLA has determined the elastic constants C_{ij} and dC_{ij}/dT as functions of T with good precision up to temperatures far in excess of the Debye temperature. Some of the results for MgO include: the accurate determination of the Grüneisen parameter to high T ; the conclusion that $(\partial C_V/\partial T)_P = 0$ at high temperature (i.e., there is no perceptible anharmonicity in specific heat); and the finding that the anharmonic parameters δ_S and δ_T are independent of T at high T .

H9. Warren P. Mason: Some brief encounters, some long memories. Louis R. Testardi (Physics Department, Florida State University, Tallahassee, FL 32306)

My introduction to the science of acoustics and my first meeting with Warren Mason were nearly simultaneous if not synonymous events at Bell Labs during the mid 1960s. Living in the house of the giant has left some memories and helped shape some personal views on the nature of human achievement and the diversity of greatness. I'll recount some of the ways our paths crossed over the ensuing 20 years and try to explain how his influence went beyond what I could have imagined at that first meeting.

TUESDAY AFTERNOON, 23 MAY 1989

AMPHITHEATER, 1:00 TO 5:10 P.M.

Session I. Noise III: Application of "Modern Spectral Analysis" Techniques to Noise Problems

Patricia Davies, Cochairman

Ray W. Herrick Laboratory, Purdue University, West Lafayette, Indiana 47907

J. Stewart Bolton, Cochairman

Ray W. Herrick Laboratory, Purdue University, West Lafayette, Indiana 47907

Chairman's Introduction—1:00

Invited Papers

I1. Tutorial: Signal processing methods for machinery diagnostics. Richard H. Lyon (Department of Mechanical Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139)

Conventional machinery monitoring systems rely mainly on signal energy processing methods such as power spectrum analyses in the frequency domain and envelope analysis in the time domain. Other, less well-known, techniques can be very useful in extracting diagnostic information from the vibration or sound produced from the machine as it operates. These methods include cepstral analysis as an order analysis or deconvolution procedure, and am and fm demodulation using the Hilbert transform to study variations in loading or speed of rotational devices. They also include various system modeling methods such as Kalman filtering and functional expansions of the transfer function. Illustrations of the use of these methods, and descriptions of situations in which each may be helpful will be presented in this tutorial.

12. Time-frequency spectra for nonstationary acoustic signals—The Wigner distribution, the evolutionary spectrum, the modified moving window spectrum, and their interrelationships. Jennie Moss, Jong-Sik Lee, Panos G. Adamopoulos, and Joseph K. Hammond (Institute of Sound and Vibration Research, Southampton University, Southampton SO9 5NH, England)

The time-frequency description of acoustic signals is a common requirement (applications include speech, sonagrams, frequency tracking, etc.), and this applies to transients and nonstationary signals (both random and nonrandom). Three time-frequency spectra that are used are the Wigner-Ville distribution, Priestley's evolutionary spectral density, and Kodera's modification to the moving spectrogram. This paper will present: a summary of the definitions and computation of these three different time-frequency spectra; the modeling of acoustic signals due to propagating sources in a form allowing **prediction** of time-frequency spectra; development of the interrelationships between the spectra; and theoretical and simulation results from time-frequency spectra. Examples of the application of all three approaches will be illustrated by using frequency-modulated signals. The signals will model sound as perceived by an observer due to a convecting acoustic source (i.e., including Doppler, range, and directivity effects). The source signal will exhibit both tonal and random components.

2:00

13. Median adaptive filtering for spectral line enhancement. Tarek I. Haweel and Peter M. Clarkson (Department of Electrical and Computer Engineering, Illinois Institute of Technology, Chicago, IL 60616)

Adaptive line enhancement using the well-known LMS algorithm is a technique for the enhancement of low-level periodic components of unknown frequency in a background of broadband noise. The LMS filter operates by iteratively minimizing a quadratic index using an instantaneous estimate of the gradient of the quadratic performance surface to update the filter coefficients. Provided the adaptation rate is sufficiently small, the filter usually performs well. If the input data are corrupted by impulsive noise, however, the filter is badly degraded, and the adaptive process may actually fail. This paper addresses this problem by defining a median least-mean-squares (MLMS) algorithm where the instantaneous gradient estimate of the LMS is replaced by the sample median of that parameter. The MLMS filter is compared, through simulation, with the LMS and with an alternative gradient smoother that utilizes the sample mean (the ALMS algorithm). Results demonstrate that the MLMS is largely insensitive to the presence of impulsive interference and performs comparably with the LMS and ALMS when no interference is present.

2:20

14. Measurement of the equivalent point source position for an arbitrary distribution of sources. S. A. L. Glegg and J. R. Yoon (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

In many applications, knowledge of the equivalent point source position of a source distribution is required for noise control calculations. For instance, the insertion loss of a noise barrier depends on the height of the noise source above the ground, and some estimate of this must be used at the design stage. The equivalent point source position is defined as the location at which a single source would be placed to give the best representation of the acoustic field over the largest possible solid angle. This paper will describe how the equivalent point source position can be measured using parametric methods. Least-squares estimators using a microphone array are shown to be of limited value, especially at low frequencies, and often the best results are obtained using a two-microphone method. [Work sponsored by the Florida Department of Transportation.]

2:40–2:55
Break

2:55

15. Accurate spectral estimation of multiple sinusoids using an FFT. John C. Burgess (Department of Mechanical Engineering, University of Hawaii, Honolulu, HI 98622)

Accurate estimates of amplitude, frequency, and phase of multiple sinusoids in a signal can be obtained using a method described earlier [J. C. Burgess, *J. Acoust. Soc. Am. Suppl.* 1 **83**, S92 (1988)]. The FFT size required for specified spectral resolution is discussed, as well as some computational aspects.

3:10

16. Interior idle noise in current model passenger cars. Timothy J. Foulkes (Cavanaugh Tocci Associates, Inc., 327 F Boston Post Road, Sudbury, MA 01776)

The author, a consultant in architectural acoustics, became interested in the vast differences in interior noise levels between passenger cars in today's market. Octave-band noise levels were measured in 20 different automobile models. In order to reduce the number of variables, the measurements were limited to the driver's position with the car stopped and engine at idle. In the 31-Hz band, noise levels varied 20 dB between two different models by the same manufacturer. The range of A-weighted levels spanned only 8 dB. While the A-weighted sound levels as published by most consumer magazines provide some indication of relative loudness, examination of octave-band levels shows radical differences between models. Some observations and conclusions are drawn by an "enlightened" consumer.

3:25

17. Measurements of vehicle noise source height for noise barrier design. Stewart A. L. Glegg and J. R. Yoon (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

The height of a noise barrier required to give a certain insertion loss is strongly dependent on the elevation of the noise source above the ground. Currently, noise barrier design methods assume equivalent point source heights for different types of vehicles, but these have never been substantiated by measurements. This study used a microphone array, with parametric processing, to evaluate the equivalent noise source height of 100 different vehicles. The results give the noise source position as a function of frequency between 200 Hz and 2.5 kHz. The relative importance of engine, stack and tire noise on the source height is apparent. It was found that the equivalent source on heavy trucks is at 1.2 m, medium trucks 0.7 m, and small vehicles 0.6 m, above the ground. [Work sponsored by Florida Department of Transportation.]

3:40

18. Nearfield high-resolution source localization. Paul Bertrand (Office National d'Etudes et de Recherches Aéronautiques, BP 72, 92322 Châtillon Cedex, France)

An example of passive acoustic array processing consists of localizing the emanating machinery noise sources along a ship shell. When possible, an efficient way to increase the poor accuracy of this localization is to move the array Fresnel zone up to the sources. A linear beamforming with

a spherical wave model then provides source bearing, range, and amplitude estimation. However, even for large antennas, the spatial resolution of this nearfield beamforming is limited to a wavelength fraction. To increase the localization accuracy, especially for low frequencies, "high-resolution" methods should be used. This paper presents, in this case, Capon's method, also called adaptive beamforming. Its performances and resolving power are computed and validated through numerical simulations. They extend to the nearfield case the results usually found in a farfield situation. However, much more drastically than with plane waves, experimental results show that this method breaks down if the source model mismatches the wave fronts actually received. To avoid this strong model sensitivity, a robust approach, extending the spectral density matrix source subspace, is proposed. This method yields correct estimation of the amplitude and position of sources too close to be resolved by linear processing. This enhancement of Capon's method does not require an accurate source model. In addition, it has a low computational cost and is not nearfield dependent. [Work supported by DCN.]

3:55

19. Cancellation of noise in the presence of signals of short duration. Vijay Raman, Osama A. Mohammed, Kang Yen, and Kurt Ramdin (Department of Electrical Engineering, Florida International University, Miami, FL 33199)

This work focuses on the active cancellation of acoustic noise in industrial environments such as factories, airports, etc., by the use of earphones with active components. An important consideration in this problem is that undesired components (noise) should be reduced without eliminating certain desired signals. More specifically, the cancellation technique must incorporate the significant constraint that warning sounds and brief verbal exchanges must not be canceled. Achieving this goal is particularly difficult when the characteristics of the desired signals resemble that of the noise. In the course of this work, these researchers have used the fact that these sounds are of relatively short duration, whereas the ambient noise environment is persistent, to formulate and evaluate appropriate signal detection and adaptive noise-canceling schemes. The filtering algorithm adopted is a variation of the well-known adaptive least-squares algorithm. [Work supported by the Florida High Technology and Industry Council.]

4:10

110. Sound power level and directivity pattern determination of a space shuttle solid rocket booster. Thomas S. Adams (Ebasco Services, Incorporated, 145 Technology Park/Atlanta, Norcross, GA 30092)

The sound power level and directivity pattern of sound were determined for a space shuttle solid rocket booster motor during a horizontal test firing at the Morton Thiokol Wasatch Test Facility in Utah. Six Bruel & Kjaer type 2230 precision sound level meters were placed at 60-deg intervals around the rocket motor at a distance of 1000 ft, except the location directly behind the nozzle (2600 ft). These analyzers were used to determine the overall sound-pressure level, between 20 and 20 000 Hz, at each of the six locations during the test. Two Cetec-Ivie model IE-30A audio spectrum analyzers were used to determine the shape of the spectrum at a one-third octave band resolution. The equipment was turned on approximately 2 h prior to the scheduled test and was unattended during the test. Various adjustments were made to the data to account for topographic and atmospheric effects. The sound produced was predominantly low frequency, and the directivity pattern exhibited major acoustic lobes 60 deg to both sides of the line of fire. A total overall sound power level of 196 dB was determined. [Work funded by NASA Stennis Space Center, Mississippi.]

I11. Results of a computer simulation scheme for active noise reduction. Kang Yen, Vijay Raman, Osama A. Mohammed, and Kurt Ramdin (Department of Electrical Engineering, Florida International University, Miami, FL 33199)

This paper presents the results of a computer simulation scheme for active noise minimization in a random noise-canceling headphone. The noise-canceling algorithm was developed in a separate paper by these authors. The performance of the noise-canceling algorithm has been tested via digital simulation of the actual noise field distributions within the ear canal. Farfield conditions have been assumed by the modeling environmental noise field to be due to sources far from the cancellation headset. The ear is initially modeled as an axisymmetric pipe, and the resulting field is calculated as a function of the location of noise canceling sources. The results proved helpful in determining the appropriate locations of the noise-canceling sources in order to achieve maximum noise energy reduction at a predetermined point in the pipe. [Project supported by the Florida High Technology and Industry Council.]

4:40

I12. Development of noise-suppression saws. T. I. Liu (Department of Mechanical Engineering, California State University—Sacramento, Sacramento, CA 95819)

Conventional wood-cutting saws have even-spaced teeth and therefore generate pure-tone noise in the cutting process. For the same noise level, the pure-tone noise is more annoying than a combination of frequencies. Thus an optimization technique is used to develop noise-suppression saws that have irregular spacing between adjacent teeth. The Fourier series can be used to express the cutting noise of the circular saws:

$$f(\theta) = \overline{f(\theta)} + \sum_{n=1}^{\infty} A_n \sin(n\theta_i + \phi_n),$$

where θ_i is the angle of the saw teeth, and A_n is the n th Fourier coefficient, which is a function of θ and represents the amplitude of the cutting noise. A conventional circular saw has a very high noise level when n is equal to integer multiples of the number of teeth. This high pure-tone noise level can be reduced and distributed to other n values if the saw teeth are unevenly spaced. In this research, an optimization algorithm is used so that A_n is close to being a constant. Producibility of the noise-suppression saws has been considered during design. Experiments have been conducted to verify the new design. This work is very helpful in improving the working environment of the woodworking factory.

4:55

I13. Noise measurements on machinery near a reflecting plane. J. B. Moreland and F. S. McKendree (Westinghouse R&D Center, 1310 Beulah Road, Pittsburgh, PA 15235)

Noise test specifications for large equipment often requires that the measurements be made with the equipment out-of-doors resting on the ground (usually a smooth, paved surface), and with the sound level meter microphone positioned between 1 and 2 m above the ground. Constructive and destructive interference between the sound radiated directly from the equipment and the sound reflected from the ground produces a sound field in which the variation of the A-weighted sound level with distance from the unit does not exactly follow the "inverse square law." The directional characteristics of the equipment under test further complicate the interpretation of the noise measurements, so that using sound levels at one distance, together with the inverse square law to predict the sound levels at other distances, will not be very precise. This paper discusses the problem of machinery noise measurement made above a reflecting plane and how modern spectral analysis techniques can be used to extract information from the noise data.

TUESDAY AFTERNOON, 23 MAY 1989

COMSTOCK A, 1:00 TO 4:45 P.M.

Session J. Psychological Acoustics II and Speech Communication II: Hearing Impairment and Sensory Aids

David A. Fabry, Chairman

Walter Reed Army Medical Center, Army Audiology and Speech Center, Washington, DC 20307

Contributed Papers

1:00

J1. Discrimination abilities of impaired listeners compared to the range of variation in performance of normal listeners. Blas Espinoza-Varas, Charles S. Watson, and Suzanne E. Patterson (Department of Speech and Hearing, Indiana University, Bloomington, IN 47405)

Discrimination abilities of sensorineural hearing-impaired listeners have been compared in most studies to average performance of small samples ($n \approx 10$) of selected, highly trained, normal listeners. Such comparisons may be biased because: (a) They do not take into account the broad distribution of the performance of normal listeners, that is, the large intersubject differences exhibited by normal listeners on many auditory

tasks; (b) unusually poor performers have been excluded from samples of normal listeners; and (c) normal listeners are often college students, impaired listeners are typically more heterogeneous samples. In the present study discrimination performance of 23 impaired listeners was compared to the range of discrimination performance exhibited by large ($n > 100$), fairly heterogeneous samples of untrained normal listeners. The tasks included discrimination of frequency, duration, intensity, tonal patterns, and nonsense syllables, and were presented at sufficiently high sensation levels (>25 – 30 dB SL). The range of normal performance was defined as the interval between the average psychometric function of the listeners comprising the best 10% of the normal sample and the function for the worst 10%. Performance of normal listeners was distributed over broad

ranges in all tasks. Performance of impaired listeners was clearly bracketed by the range of normal performance in all of the nonspeech discrimination tasks. Variability of impaired listeners' performance in the nonspeech tasks was comparable to that of normal listeners, and their psychometric functions were uniformly distributed over the range of those for normal performance.

1:15

J2. Use of the Articulation Index to assess "noise reduction" hearing aids. David A. Fabry (Walter Reed Army Medical Center, Washington, DC 20307) and Dianne J. Van Tasell (University of Minnesota, Minneapolis, MN 55455)

The Articulation Index (AI) was used to evaluate a commercially available "noise reduction" hearing aid. First, a transfer function relating calculated AI to rated speech intelligibility was derived empirically for a group of normal-hearing subjects. This function was used to predict speech recognition by 12 hearing-impaired subjects wearing either the noise reduction hearing aid or conventional amplification. Subsequently, subjects rated the intelligibility of connected discourse presented under both aided conditions at five signal-to-noise (S/N) ratios in speech-weighted noise. The AI transfer function predicted speech intelligibility accurately for half the subjects. For the remaining subjects, predicted scores were typically higher than observed scores. If the effects of upward spread of masking were included in the AI calculations, these differences were minimized. For all subjects, speech intelligibility increased monotonically with AI. AI scores and speech intelligibility ratings were similar between the two hearing-aid conditions, suggesting that S/N ratio was not increased by the noise reduction hearing aid relative to conventional amplification. [Work supported by NINCDS NS 12125.]

1:30

J3. A different approach to the noise/interference problem of the hearing impaired. Edgar Villchur (Foundation for Hearing Aid Research, P.O. Box 306, Woodstock, NY 12498)

Speech has been called an error-resistant code because it contains many redundant cues to its meaning. These cues provide a reserve against the loss of cues masked by interference and allow normal listeners to understand speech in noisy environments. Evidence from experiments with processed speech suggests that the special vulnerability of the hearing impaired to interference is not an intrinsic feature of the impairment but typically derives from a loss of perception of the redundant cues enjoyed by normal listeners. If such is the case, restoring lost speech cues to the impaired listener is likely to be more helpful against interference than increasing signal-to-interference ratios. The lecture will include a demonstration of speech processed to simulate the effects of recruitment and accentuated high-frequency loss, and the vulnerability of this speech to interference. Lost cues will then be restored by compensatory processing (two-channel amplitude compression plus frequency shaping), which increases the intelligibility of the speech in the presence of interference even though it decreases the signal-to-interference ratio.

1:45

J4. On the difficulty of fricative perception by hearing-impaired subjects. Fan-Gang Zeng and Christopher W. Turner (Communication Sciences and Disorders, Syracuse University, Syracuse, NY 13244-2280)

By comparing conditions under which both normal and hearing-impaired subjects were presented with equivalent degrees of audibility for fricative perception, the current study isolated the factors of lack of audibility from that of loss of suprathreshold discriminability of fricative

perception cues. One moderate and two severe cochlear-impaired subjects identified synthetic fricatives, with either full cues (frication and transition) or frication cue alone, presented in closed-set response tasks across a wide range of presentation levels. The results showed that hearing-impaired subjects yielded equivalent recognition performance to the normals when given an equivalent degree of audibility for the frication cue, but they obtained poorer-than-normal performance if only given an equivalent degree of audibility for the transition cue. Considering the 20- to 25-dB intensity difference between the frication and transition cues, the difficulty that hearing-impaired subjects have in perceiving fricatives under normal circumstances may be due to two factors: the lack of audibility in terms of frication cue and the loss of discriminability in terms of transition cue. [Work supported by SU Senate Research Grant and Deafness Research Foundation.]

2:00

J5. A comparison of speech discrimination with cochlear implants and tactile aids. A. E. Carney (Boys Town National Institute, Omaha, NE 68131), M. J. Osberger, A. Robbins, J. Renshaw, and R. Miyamoto (Indiana University School of Medicine, Indianapolis, IN 46223)

Speech discrimination was assessed in three groups of hearing-impaired children fitted with three different sensory aids: (1) single-channel House 3M cochlear implants; (2) 22-channel Nucleus cochlear implants; and (3) Tactaid II devices. Nine subtests of a speech discrimination battery were constructed, which focused both on suprasegmental and segmental speech contrasts. A change/no change task was used to assess discrimination. A change trial consisted of four standard stimuli followed by four contrast stimuli; the no-change trial had eight standard stimuli. Hits and false alarms were tallied, and $P(C)_{\max}$ was computed for each subject for each subtest. Results indicated that, in general, subjects with Nucleus devices had the highest mean $P(C)_{\max}$ across subtests, followed by subjects with 3M House and Tactaid II devices. However, intersubject variability was high for each device. In addition, differences between devices frequently varied on individual subtests. The ability of this test battery to reveal differences between children's speech perception with different sensory aids will be discussed. [Work supported by NIH (NS24875).]

2:15

J6. Two-channel FFT techniques for measuring hearing aids with adaptive signal processing. John R. Bareham (Brüel & Kjaer Instruments, Incorporated, 185 Forest Street, Marlborough, MA 01752)

Hearing-aid measurements made in test enclosures have traditionally utilized a sinusoidal method such as that of ANSI Standard S3.22-1987. While this method works well with essentially linear aids, it gives misleading results with automatic gain control (AGC) aids. A new method using shaped noise has been under study in Working Group S3-48. This works well with aids having linear, AGC, or adaptive frequency response characteristics, but does not work with aids having speech versus noise discrimination based on detection of the modulation of the received signal. This paper proposes an extension of the shaped noise method that seems to work well with aids having modulation detection. The proposed method uses a combination of pulsed, shaped noise and continuous bandlimited noise as the test signal. The analysis is by two-channel FFT methods. Examples of appropriate and inappropriate measurement methods will be shown for all types of aids mentioned above.

J7. Implementation of principal components spectral analysis of speech in a wearable tactile aid. T. J. Rahrer, T. A. Kwasniewski (Department of Electronics, Carleton University, Ottawa, Ontario K1S 5B6, Canada), and B. W. Tansley (Department of Psychology and Department of Systems and Computer Engineering, Carleton University, Ottawa, Ontario K1S 5B6, Canada)

One common configuration of a vibrotactile sensory aid for the hearing impaired is the vocoder scheme employing a bank of bandpass filters that model human audition. These systems result in 16 or more spectral

energy outputs, each of which drives a vibrator. Although successful in laboratory and classroom settings, the large number of vibrators required makes the implementation of wearable versions difficult. By performing a principal components analysis of the filter bank outputs, the spectral information can be recorded in as few as four parameters. A compact principal components analysis system has been implemented using digital signal-processing techniques. The system consists of a 1/3-oct bandpass filter bank, high-speed energy detectors for each filter, logarithmic scaling of spectral energies, and a principal component calculator, all in software running on a commercial DSP microchip. In this paper, the principal components system is compared to the traditional vocoder method, as well as to the pitch and formant extraction schemes.

2:45-3:00
Break

3:00

J8. Recognition of Thai consonants by listeners with normal hearing and noise-induced hearing loss. Sumalai Maroonroge (11510 Ridge Run Drive, Houston, TX 77064), Igor V. Nabelek (Department of Audiology and Speech Pathology, University of Tennessee, Knoxville, TN 36996-0740), and Anothai U-phanich (Speech and Hearing Clinic, Ramathibodi Hospital, Bangkok 10100, Thailand)

The purpose of this study was to compare recognition of Thai consonants in listeners with normal and noise-induced hearing loss. The subjects consisted of 36 normal-hearing and 120 hearing-impaired listeners whose hearing loss starting at 3, 4, or 6 kHz. All subjects were native Thais. The test materials were meaningful Thai monosyllabic words recorded by a female speaker. The test was presented at 35 dB SL in quiet and in noise (noise level was 10 dB below signal level). There was a significant difference between scores obtained by listeners with normal hearing and those with noise-induced hearing loss. Consonant recognition scores in quiet and in noise were also different. The overall scores from the three groups of noise-induced listeners were similar but the patterns of consonant recognition were distinct for each group and test condition. Recognition scores for stops, fricatives, and affricates were relatively low. Aspirated consonants showed lower scores than unaspirated sounds. Confusion errors occurred across both place and manner of productions.

3:15

J9. Simulation of sensorineural hearing loss. Paul Duchnowski and Patrick M. Zurek (MIT Research Laboratory of Electronics, Room 36-761, 50 Vassar Street, Cambridge, MA 02139)

Previous work [Zurek and Delhorne, *J. Acoust. Soc. Am.* **82**, 1548-1559 (1987)] found that noise masking, while degrading the speech reception of normal listeners to resemble closely that of the impaired listeners, could only be used to simulate losses less than 70 dB. With greater losses, the noise is unacceptably intense. In the present simulation, a variation of Villchur's [*J. Acoust. Soc. Am.* **62**, 665-674 (1977)] system, the input signal is divided into 14 1/3-oct bands and each band level (determined over a 20-ms window) is compared to the impaired threshold at the center frequency of the band. For input levels below threshold, the signal is attenuated to essentially zero. Signals in an X -dB range ($X \approx 30$ dB) immediately above threshold are attenuated according to an expansive mapping of input level into the range between normal threshold and impaired threshold plus X dB. Signals more than X dB above the impaired threshold are passed unattenuated. The system thus simulates the two primary characteristics of sensorineural hearing loss—elevated threshold and recruitment. In pilot studies, this technique was used successfully to simulate consonant reception of impaired listeners described by Zurek and Delhorne. To evaluate the simulation further, new subjects (both impaired and normal) are being tested with both syllables and sentences with variations in overall level, signal-to-noise ratio, and frequency-gain characteristic. Intelligibility scores and SINFA analyses obtained thus far

show this processing to be as good as additive noise in simulating sensorineural hearing loss. [Work supported by NIH.]

3:30

J10. Modeling a vented push-pull hearing-aid fitting *in situ* as a feedback control system. David A. Preves and James R. Newton (Argosy Electronics, 10300 W. 70th Street, Eden Prairie, MN 55344)

For the investigation of acoustic feedback encountered with push-pull amplification for in-the-ear hearing aids, a simulated fitting was considered a feedback control system to include the acoustic leakage from the hearing-aid receiver traveling back to the microphone. A Gennum WS531 amplifier was connected to an earmold shell made for the right ear of KEMAR. This shell contained a Knowles EK 3024 microphone, a Knowles BK1613 receiver, and a 0.09-in.-diam by 0.8-in.-long vent hole. Transfer function magnitude and phase were obtained for the microphone, amplifier, receiver, for the open loop while breaking the signal path between microphone and amplifier, and for the closed loop, all with maximum gain before onset of oscillation. The acoustic feedback transfer function was determined by subtracting the forward loop transfer function from the closed loop transfer function. A circuit synthesis program was used to fit the transfer function measurement data into mathematical transfer function expressions. Comparisons were made between the synthesized transfer functions and the measured transfer functions.

3:45

J11. Reducing the effects of target misalignment in an adaptive beamformer for hearing aids. Julie E. Greenberg, Patrick M. Zurek (MIT Research Laboratory of Electronics, Room 36-761, Cambridge, MA 02139), and Patrick M. Peterson (BBN Laboratories, 10 Moulton Street, Cambridge, MA 02138)

Previous work [Peterson *et al.*, *Acta Otolaryngol.* (in press)] has shown that a multimicrophone adaptive beamforming system for hearing aids reduced noise effectively at input target-to-jammer ratios (TJRs) up to about 0 dB. However, that beamforming algorithm assumed that the target signal was identical at the microphones, and violation of this assumption caused degrading target cancellation when input TJR was greater than 0 dB. The present system uses a measure of intermicrophone correlation to estimate short-time TJR and inhibits adaptation dynamically when TJR exceeds a selected threshold. Computer simulations of a two-microphone system in anechoic and moderately reverberant environments show that this method reduces (speech) target cancellation when the identical target assumption is violated. The improved system reduces noise at both positive and negative input TJRs (± 20 dB), with target misalignments up to 10 deg. Additionally, this processing scheme reduces the misadjustment noise caused by strong target signals in the adaptive feedback loop. [Work supported by NIH.]

J12. Effect of number of channels on the performance of a multichannel compression hearing aid. E. William Yund and Krista M. Buckles (Hearing Loss Research Laboratory, Veterans Administration Medical Center, Martinez, CA 94553)

Previous research [E. W. Yund, H. J. Simon, and R. Efron, *J. Rehab. Res. Dev.* **24**, 161-180 (1987)] has demonstrated the effectiveness of an eight-channel compression hearing aid for individuals with sensorineural hearing loss in the task of speech discrimination in a background of speech-band noise. In the present study, similar multichannel compression systems with 4, 6, 8, 12, and 16 channels were compared with each other and with linear (frequency equalized amplification) and unprocessed (flat amplification) conditions. The results on 15 hearing loss subjects showed increasing speech discrimination performance as the number of channels increased, although the difference between 4 and 8 channels was much greater than the difference between 8 and 16 channels. Confirming the previous results, the performance advantage of the multichannel compression conditions over the linear and unprocessed conditions was greater at low signal-to-noise ratios (S/N), e.g., +5, 0, or -5 dB as opposed to +10 or +15 dB. [Work supported by VA Rehabilitation Research and Development Service.]

4:15

J13. Real-time cepstral analysis, speech feature extraction, for the cochlear prosthesis. Ibrahim J. Gedeon, Tad A. Kwasniewski, and Timothy J. Rahrer (Department of Electronics, Carleton University, 1231 Colonel By Drive, Ottawa, Ontario K1S 5B6, Canada)

The most common speech-processing strategies for use in the cochlear implant employ multifeature extraction. For the proposed application, the first three formant frequencies and amplitudes along with the pitch

period are calculated for each speech frame processed. This paper presents real-time cepstral analysis as a complete speech analysis strategy for the cochlear implant. Cepstral analysis has long been known as an accurate method for pitch extraction and formant tracking. However, it has not been previously considered for real-time signal-processing applications, due to the complexity of the calculations involved. Taking advantage of today's microprocessor technology, cepstral analysis has been implemented using the Texas Instruments TMS320C25 digital signal processor. Results obtained with a 10-pole LPC analysis using the same processor are compared with those of the proposed method. The results are also compared to a commercial software package (ILS from STI) for validation.

4:30

J14. Some factors affecting recovery from poststimulatory fatigue. I. M. Young and L. D. Lowry (Department of Otolaryngology, Jefferson Medical College of Thomas Jefferson University, Philadelphia, PA 19107)

Recovery from poststimulatory fatigue was measured on subjects with normal hearing and patients with unilateral cochlear and retrocochlear hearing impairments. Ears were stimulated by steady tones at various intensity levels for 3 min. Thresholds during the recovery period were obtained at the stimulated frequencies by using both the steady tone only and the tones alternated between interrupted and steady tones. For all subjects, alternate presentation of tones resulted in more rapid recovery to preexposure level than steady tone presentation. These results seem to indicate that the off-time of interrupted tones permits the fast recovery from fatigue for the auditory nervous system. It is presumed that, by varying the on-off ratio, a variety of recovery patterns may be measured.

TUESDAY AFTERNOON, 23 MAY 1989

NEWHOUSE II, ROOM 254, 1:00 TO 2:24 P.M.

Session K. Speech Communication III: Timing and Stress in Phonemes and Syllables

Robert J. Porter, Jr., Chairman

Kresge Hearing Research Laboratories, 1100 Florida Avenue, New Orleans, Louisiana 70119

Contributed Papers

1:00

K1. Final lengthening: A local tempo change. Jan Edwards (Hunter School of Health Sciences, 425 East 25 Street, New York, NY 10010), Mary E. Beckman, and Janet Fletcher (Ohio State University, Department of Linguistics, 204 Cunz Hall, 1841 Millikin Road, Columbus, OH 43210)

This paper compares the articulatory correlates of lengthening at phrase boundaries to those of slowing down from normal to slow tempo. The velocity, displacement, and duration of opening and closing mandibular gestures of syllables in intonation-phrase-final and nonfinal position were examined for four speakers. The longer durations of phrase-final syllables are due primarily to slower closing gestures, without any accom-

panying differences in articulator displacement. Similarly, changing from normal to slow tempo is effected by a decrease in peak velocity for both the opening and closing gestures, without any accompanying differences in articulator displacement. Thus final lengthening can be described as a local slowing down before the phrase boundary. Interestingly, two subjects did not differentiate between the peak velocity of final and nonfinal closing gestures at the slow rate, probably because they had reached a lower limit on peak velocity. However, these subjects did maintain a difference in vowel period duration between nonfinal and final syllables at the slow tempo. This result suggests that final lengthening has a directly temporal target at a more abstract level of specification than the articulatory dynamics. [Work supported by the National Science Foundation under grants IRI-8617873 and IRI-8617852.]

K2. Vowel length and closure duration in word-medial VC sequences. Stuart Davis and W. Van Summers (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Previous research has shown that English vowel length varies depending on the voicing characteristic of the following consonant. For stop consonants, closure durations also vary as a function of consonantal voicing. Generally, vowel-stop sequences containing voiced consonants show longer vowel durations and shorter closure durations than similar sequences containing voiceless consonants. These previous studies have focused on stressed vowels in monosyllabic or bisyllabic words. Very little research has examined the effects of postvocalic voicing on stressless vowels. In the present study, the influence of postvocalic voicing on vowel and closure durations in VCV and VCV sequences is studied. Subjects produced sentence pairs containing target words contrasting in intervocalic consonantal voicing (e.g., adopt-atop, tabbing-tapping). Both stressed and unstressed vowels tended to lengthen before voiced consonants. However, the vowel-lengthening effect was not as consistent for stressless vowels as for stressed vowels. Closure durations were longer for voiceless stops than voiced stops after a stressed vowel. However, voicing effects on closure duration were inconsistent after stressless vowels. The results have implications concerning perceptual cues for intervocalic voicing and for issues concerning syllable-internal structure. [Work supported by NIH.]

1:24

K3. Articulatory consonant and vowel timing in rhythmically produced syllables. Barbara Kühnert, Bernd Pompino-Marschall, and Hans G. Tillmann (Institut für Phonetik und Sprachliche Kommunikation, Ludwig-Maximilians-Universität, Schellingstrasse 3, D-8000 Munich 40, Federal Republic of Germany)

The results of experiments using simultaneous EMG recordings for consonantal (OOS) and vocalic gestures (ABD) in rhythmically produced speech are discussed in connection with an articulatorily based interpretation of the *P*-center phenomenon. Three subjects produced sequences of five monosyllabic tokens of the form /Cak/ (with /C/ = /p/, /pf/, /pl/, /pr/, /pfl/) in time to a metronome. In contradiction to the hypothesis of Tuller and Fowler [Percept. Psychophys. 27, 277-283 (1980)], the data in this study suggest that the timing of the articulatory vocalic gesture is not a direct correlate of the *P* center and, furthermore, not independent of the syllable-initial consonants. In addition, although the acoustic syllable onset shows the well-known systematic deviations from isochrony depending on the length of the initial consonant cluster, the articulatory consonantal gesture clearly does not show a parallel systematic shift to earlier onset. These results support the hypothesis that the articulatory timing pattern of monosyllabic utterances seems to be determined on the basis of the phonetic structure of the entire syllable.

1:36

K4. *P* centers, *C* centers, or what else? Bernd Pompino-Marschall, Barbara Kühnert, and Hans G. Tillmann (Institut für Phonetik und Sprachliche Kommunikation, Ludwig-Maximilians-Universität, Schellingstrasse 3, D-8000 Munich 40, Federal Republic of Germany)

Some questions concerning the underlying nature of the so-called *P*-center phenomenon (the acoustic anisochrony of rhythmically regularly produced/perceived sequences of monosyllables) are discussed in the light of recent EMG data from experiments with rhythmically produced /Cak/ syllables (with /C/ = /p/, /pf/, /pl/, /pr/, /pfl/). The differences between /p/ vs /pf/ syllables resemble the timing differences in the microbeam data of Browman and Goldstein [Haskins SR-93/94, 85-102 (1988)]. Whereas they found a stable timing relation between the "articulatory mean" of the initial consonants (termed *C* center) and the

acoustical offset of the following vowel, in terms of their task dynamic description, however, the phase relations between consonant and vowel gestures should remain the same. This would result in a reduced stiffness of the vowel gesture for the syllables containing consonant clusters. The observations of the present study would also correspond to a reduction of stiffness for the vowel but the numerical data seem to contradict a stable phasing relation. In connection with the psychoacoustic interpretation of the *P*-center phenomenon, an argument is made against a strictly intrinsic timing interpretation (based on articulatory regularities) of the effects of rhythmical production/perception.

1:48

K5. Universal versus language-specific aspects of voicing-dependent vowel duration. Christiane Laeuffer (Romance Languages, Ohio State University, 248 Cunz Hall, Columbus, OH 43210)

All languages show vowels to be slightly longer before voiced obstruents. A phonological rule of vowel lengthening has been proposed for English because the variation seems to be much greater than in, say, French. This comparison, however, has been based on studies that do not control for word length, syllable structure, or for whether or not the languages have final devoicing. A comparison of voicing-dependent vowel duration in English and French monosyllables shows that phrase finally and medially under focus, obstruent voicing is a significant factor influencing the duration of preceding vowels in *both* languages. In unfocused medial position, on the other hand, the difference between the durations as a function of consonant voicing is much smaller. Moreover, if differences in syllabification are taken into account, the languages are even more similar. That is, in French, consonants tend to be resyllabified with the following syllable, unlike in English. These results suggest that languages universally exhibit fairly similar physiologically conditioned voicing-dependent variation in vowel durations. Under certain conditions, including word length, syllabic structure, stress, position in word, and utterance and speaking tempo, the difference is enhanced, while in others it is obscured. Thus a general phonetic account in terms of timing control is preferable to a language-specific phonological rule.

2:00

K6. An articulatory study of consonant-vowel overlap. Kenneth de Jong (Department of Linguistics, Ohio State University, Columbus, OH 43210)

Fowler [Phonetica (1981); J. Exp. Psychol. (1983)] has proposed that speech production is a series of global articulatory movements associated with the vowels of an utterance, upon which smaller, consonantal movements are superimposed. A possible articulatory test of this theory models consonants and vowels as alternations between a relatively closed and open vocal tract. Using the Wisconsin x-ray microbeam system, articulatory records were obtained of English phrases with varied accent placement and alveolar consonant sequences of varied duration. Analysis of one speaker's productions is now complete. When the data are pooled over all conditions, the total duration of the bisyllabic sequence and the time difference between sonority peaks is strongly related to the length of the medial consonants; there is no evidence for articulatory overlap—i.e., vowel shortening in the neighborhood of long consonants or shifting of sonority peaks toward longer consonants. However, correlations within accent types uncover a strong shortening effect for accented syllables preceding long consonants, along with a reduced displacement in the following vowel. Thus any comprehensive overlapping consonant-vowel theory of speech production must take into account the general prosodic structure of the utterance. [Work supported by the National Science Foundation under grant No. IRI-8617852 to Mary Beckman.]

K7. The timing of lip rounding and tongue backing for /u/. Gina M. Lee (Department of Linguistics, Ohio State University, 204 Cunz Hall, 1841 Millikin Road, Columbus, OH 43210)

A small corpus of x-ray microbeam data was examined to test the predictions made by two well-known views of anticipatory coarticulation: time locking and feature spreading. Lip rounding and tongue backing associated with English /u/ were investigated. The token types were VC_nV sequences where V = /i, u/ and C = /s, t/, and consonants are

assumed to be neutral with respect to both features under study. The speaker showed no evidence of time locking or feature spreading of the onset of lip rounding in /iC_nu/-type tokens. Within the same token types, there was an absence of tongue backing activity during the production of the consonants. However, other articulatory phenomena are temporally fixed. The maximum degree of rounding was fixed relative to the acoustic onset of /u/ [as stated in G. M. Lee, *J. Acoust. Soc. Am. Suppl.* 1 81, S37 (1987)]. This is consistent with past findings involving EMG peaks [Hirase *et al.*, *Ann. Bull. RILP* (1968, 1969)]. The timing of the minimum degree of backing was also fixed. These findings may provide evidence for the notion of a (static) target in speech production.

TUESDAY AFTERNOON, 23 MAY 1989

NEWHOUSE I, ROOM A1, 1:15 TO 5:00 P.M.

Session L. Bioresponse to Vibration I and Physiological Acoustics II: Structural and Functional Aspects of Mechanotransduction

Stanley J. Bolanowski, Jr., Chairman

Institute for Sensory Research, Syracuse University, Syracuse, New York 13244

Invited Papers

1:15

L1. Primary and secondary mechanoreceptors. S. J. Bolanowski (Institute for Sensory Research, Syracuse University, Syracuse, NY 13244)

A brief review of the structure and function of primary (i.e., lacking a synapse) and secondary (i.e., having a synapse) mechanoreceptors will be presented. The central theme of this review and the session in general will be "What can be learned about mechanotransduction in general, based on the similarities and differences of the two receptor types?"

1:30

L2. Stretch-activated ion channels as acoustical transducers. Frederick Sachs (SUNY Department of Biophysical Sciences, Buffalo, NY 14214)

Ion channels sensitive to membrane distortion have been demonstrated in nearly all cells, from plants to animals, in specialized sensory transducers, such as the crayfish stretch-receptor and, in traditional motor outputs, such as skeletal muscle. The channels respond to membrane tension by opening with a probability that (for low tensions) follows an exponential in the square of tension. The data can be explained with a simple model in which tension does Hookean work on the channel raising its energy, which leads towards transitions to the open state. Even in skeletal muscle, the channels are sufficiently sensitive to tension to serve as productive models of hair cell transduction. If auditory transduction is due to stretch sensitive ion channels in ciliary membranes, a very simple traditional model of force coupling between stereocilia leads to the following predictions [F. Sachs, "Biophysics of mechanoreception," *Membrane Biochemistry* 6 (2), 173-195 (1986)]: (1) tufts are tapered in order to provide a region of strain that is first order in the angle of deflection; (2) channels may be located over the cilia, but will only be turned on at the distal tips in the strained region of membrane; (3) for maximum sensitivity, the optimum number of channels is on the order of one/tip; (4) given a constant channel density, to maintain optimum sensitivity at different mean pressures requires some form of mechanical relaxation to the optimal tension; (5) specialized tip links connected to channels are not required since strain is applied in the plane of the membrane; and (6) the channels are sufficiently fast to account for the response time of the generator current and the transduction system is sufficiently sensitive to account for the observed psychophysical "thresholds." [Work supported by NIADDK-13195 and USARO 22560-LS.]

L3. Localization and biophysical characterization of an adaptation mechanism in amphibian hair cells. John A. Assad, Nir Hacohen, and David P. Corey (Department of Neurology, Massachusetts General Hospital, Boston, MA 02114 and Howard Hughes Medical Institute, Boston, MA 02114)

Hair cells of the bullfrog sacculus adapt to maintained displacement stimuli in a manner that indicates an active movement of the elastic element that exerts stress on transduction channels. Transduction and adaptation were present in dissociated hair cells studied with whole-cell, patch-clamp recording and video microscopy. The adaptation rate—as well as the position of the resting current-displacement $[I(X)]$ curve—depended on extracellular calcium and on membrane potential, suggesting that calcium passes into the cytoplasm to reach its site of action. Following a step hyperpolarization, the adaptation increased within milliseconds, suggesting that the calcium site is within a few microns of the ion channels through which calcium enters. The voltage dependence of the resting $I(X)$ curve, together with the “gating springs” hypothesis for transduction, predicts a 40-nm movement of the bundle away from the kinocilium when the cell is depolarized. This was observed.

L4. Actin in cochlear outer hair cells: A structural basis for cell shape and motility. Norma Slepecky (Institute for Sensory Research, Syracuse University, Syracuse, NY 13244-5290)

Threshold sensitivity and frequency selectivity in the cochlea depend on intact outer hair cells and models to explain these phenomena suggest that active mechanical processes, residing in outer hair cells, alter micro-mechanical properties of the basilar membrane. Auditory sensory hair cells isolated from the inner ear change their shape in response to stimulation *in vitro*. Three areas of the cell have been suggested to be involved in this form of cell motility: (1) the infracuticular network of actin filaments that is seen as a central core; (2) the cortical layer of actin filaments along the lateral wall; and (3) actin in the outer hair cell cytoplasm. The actin filaments in the stereocilia, the cuticular plate, and the infracuticular network are not involved, since mechanically disrupted cells lacking these structures still shorten. Actin filaments along the lateral wall are not involved in shortening since outer hair cell membrane ghosts do not shorten, however, they retain the ability to constrict so the cell becomes thinner. Shortening of outer hair cells does occur in mechanically disrupted cells as long as they contain some cytoplasm. Thus shortening of outer hair cells requires cytoplasmic components. Immunocytochemical studies have demonstrated the presence of microtubules, spectrin, calcium binding proteins, and actin in the outer hair cell. Staining with phalloidin indicates for the first time that actin is present as short filaments, and, with transmission electron microscopy, filaments are seen as a network throughout the cytoplasm. Thus actin may provide the structural basis for maintaining cell shape and for causing motility.

L5. A model for mechanotransduction in a Pacinian corpuscle. Mark H. Holmes (Department of Mathematical Sciences, RPI, Troy, NY 12180) and Jon Bell (Department of Mathematics, SUNY at Buffalo, Buffalo, NY 14214)

A Pacinian corpuscle (PC) is a rapidly adapting mechanoreceptor that resides in the skin and other tissues. The transduction process in the PC consists of a mechanical filtering by the capsule that surrounds the dendrite, along with a receptor membrane component. A model is described that accounts for these. The layer structure of the capsule is taken as formed from concentric elastic membranes with the interlamellar spaces filled with a viscous incompressible fluid. From this the hoop strain in the receptor membrane is found and this serves as the input to a receptor model, which has a strain-sensitive activation mechanism for the transducer conductance. The transient behavior of the receptor potential due to various compression stimuli to the capsule is investigated. Of particular interest is how the viscoelastic properties of the capsule contribute to the adaptive behavior of the PC. To this end the dependence of the hoop strain on the rate of compression and the imposed strain are discussed along with the response of the PC when the loading is removed.

3:45

L6. Effects of static indentation on the input-output function of Pacinian corpuscles. S. J. Bolanowski, Jr. and C. M. Checkosky (Institute for Sensory Research, Syracuse University, Syracuse, NY 13244 and Department of Surgery, University of Rochester Medical School, Rochester, NY 14642)

The receptor potential (RP) of cat Pacinian corpuscles (PC) in response to sinusoidal displacements exhibit asymmetric, full-wave rectification. Furthermore, there are apparently two populations of PC, one more sensitive to stimulus compression, the other to withdrawal. These results can be explained by proposing that there are two groups of neutral elements within PC, each producing a half-wave rectified response but summing in polarity opposition. The RP may thus be dependent upon the relative contribution from the two groups. If true, then changes in the RP may occur with selective (de) activation of one group of elements. This may be tested by varying the static indentation of the vibrating probe, noting that small static indentations may permit only the most sensitive group of elements to be excited, revealing a half-wave rectified RP. Increasing indentations should then produce concomitant changes in the RP shape as the other group becomes activated. The experiments show that changes in static indentation produces changes in the RP, the changes considerably more complicated than the hypothesis predicts. The results indicate that while the two-group hypothesis may still be correct, the I-O function of each element may not resemble a half-wave rectifier. [Work supported by NS 23933.]

4:00

L7. Bidirectional transduction by haircells. C. L. Searle, H. E. Secker-Walker, T. A. Wilson, and D. M. Freeman (Room 20C-014, MIT, Cambridge, MA 02139)

Given the two-port description of a hair cell from Weiss [Hear. Res. 7, 353-360 (1982)], with the Davis model for the angle-dependent conductance representing the transduction channels [Leong and Weiss, Hear. Res. 20, 175-195 (1985)], and the data of Crawford and Fettiplace [J. Physiol. 364, 359-379 (1985)], Howard and Hudspeth [Neuron 1, 189-199 (1988)], and Holton and Hudspeth [J. Physiol. 375, 195-227 (1986)], analysis shows that (1) the feedback inherent in the two-port must be positive, and (2) degenerative (outward-flowing) channels must be present. The static nonlinear behavior of the system, and the small and large signal dynamics, have been studied. The system displays Q multiplication at low signal levels, bandwidth increase with signal level, a compressive nonlinearity at CF, and continuous oscillations if the parameters are slightly misadjusted—all well-documented physiological phenomena.

4:15

L8. Mechanical purification and protein constituents of amphibian hair cell stereocilia. Gordon M. G. Shepherd, Barbara A. Barres, and David P. Corey (Department of Neurology, Massachusetts General Hospital, Boston, MA 02114 and Howard Hughes Medical Institute, Boston, MA 02114)

In order to understand the molecular basis for transduction and adaptation in hair cells, it is necessary to know the identities and relationships of proteins in the mechanically sensitive stereocilia. A rapid and specific purification procedure has been developed for stereocilia of the bullfrog sacculus, that relies on mechanical adhesion of the stereocilia to nitrocellulose paper ("bundle blot purification"). The high purity and yield of the preparation were demonstrated by light and scanning electron microscopy. Electrophoretic separation of proteins revealed 12-18 major bands. Actin stained with phalloidin and was a prominent band at 42 kD. Fim-

brin (68 kD) was identified by immunoblot. Both bands were present in cores prepared by treating stereocilia with triton to remove membrane and soluble proteins. Calmodulin (17 kD) and calbindin (28 kD) were identified by immunocytochemistry or by comigration. The electrophoretic pattern differs substantially from that of brush border microvilli, which are structurally analogous; in particular, the calmodulin-regulated, myosinlike 110-kD protein of brush border was not detected in stereocilia.

4:30

L9. Very high levels of calmodulin (CAL) in outer hair cells (OHC) of organ of Corti (OC). Kuniaki Takahashi, Isolde Thalmann, Ruediger Thalmann (Department of Otolaryngology, Washington University, St. Louis, MO 63110), Hans-Peter Zenner (Department of Otolaryngology, University of Tübingen, 7400 Tübingen, Federal Republic of Germany), and Peter Kraus (Department of Otolaryngology, University of Würzburg, 8700 Würzburg, Federal Republic of Germany)

Using immunohistochemical techniques, Slepecky *et al.* [Hear. Res. 321, 11-22 (1988)] demonstrated a selective accumulation of CAL, an ubiquitous calcium binding protein, in the inner hair cells (IHC) and OHC of the OC. By quantitative polyacrylamide gel electrophoresis, it was found that in chinchilla OC, CAL is by far the highest in OHC, where levels of 6%-8% of total protein are reached. For comparison, the highest known accumulation of CAL (in the heads of spermatozoa) amounts to 12%. CAL is considerably lower in IHC (about 1%), and is not detectable with usual protein loads in other cell types of the OC. CAL is also very low in stria vascularis and Reissner's membrane. The relative distribution of CAL in the OC is similar to that of glycogen, where the exceedingly high level of 600-mmol/kg dry weight is reached in OHC [Thalmann *et al.*, Laryngoscope 80, 1619-1645 (1970)]. This circumstantial evidence suggests that activation of phosphorylase kinase may be an important function of CAL in OHC, whereby ultimately energy for contraction is mobilized from glycogen. Most likely CAL-induced activation of myosin light chain kinase also takes place as part of the contractile process. [Work supported by NIH and DRF.]

4:45

L10. Significance of amino acid sequence in characterization of unidentified major proteins of the organ of Corti (OC). Isolde Thalmann and Ruediger Thalmann (Department of Otolaryngology, Washington University, St. Louis, MO 63110)

Previously, two low molecular weight, highly acidic proteins were described that are present at very high concentrations in the OC and that are assigned the tentative names OCP-1 and OCP-2 [Thalmann *et al.*, Arch. Otorhinolaryngol. 226, 123-128 (1980)]. Now amino acid sequencing studies of these proteins have been initiated. Preliminary results show that the N-terminal sequence of OCP-2 is Pro-Gly-Ile-Lys-Leu-Gln-Ser-Ser-Asp-Gly-Glu-Ile-Phe-Glu-Val-Asp-X-X-Ile-Ala-Lys-Gln-. Among the advantages of knowing the sequence or primary structure are (1) the primary structure contains all the information required for formation of the secondary and tertiary structure; (2) comparison with sequences of known proteins allows determination of the degree of homology, to obtain clues about evolutionary history and degree of conservation, and hints about protein function (e.g., the "EF hand" in some calcium-binding proteins); (3) repeating motifs suggest structural role; (4) internal homology suggests duplication (e.g., in calmodulin); (5) the nature of the N-terminal residue gives clues about the turnover of the protein; and (6) the amino acid sequence allows oligopeptides to be synthesized for obtaining antibodies for immunohistochemical studies and corresponding oligonucleotides to be synthesized for molecular biologic studies. [Work supported by NIH and DRF.]

Session M. Musical Acoustics I: Acoustics of Percussion Musical Instruments

Uwe J. Hansen, Chairman

*Department of Physics, Indiana State University, Terre Haute, Indiana 47803**Invited Papers***1:30****M1. Vibration modes and tuning of xylophone bars.** Ingolf Bork (Physikalisch-Technische Bundesanstalt, D-3300, Braunschweig, West Germany)

The characteristic quality of xylophone sound depends both on the inharmonic structure of its spectrum of partials and on the order in which the individual partials decay. Subjective tests have shown that a systematic tuning of the first three partials improves the quality. A procedure for the practical realization of such a tuning is described. With the help of modal analysis, it can be shown that when torsional vibration modes as well as bending modes contribute to sound radiation, the tuning becomes more complicated.

2:00**M2. Pitch distribution in bell chimes of the Chinese Bronze Age.** Lothar von Falkenhausen (Department of Asian Languages, Stanford University, Stanford, CA 94305-2034)

Pitch measurements on a number of archaeologically discovered graduated sets of two-pitch bells [cf. M. Chengyuan, *Chin. Music* 3(4), 81–86 (1980); 4(1), 18–20 (1981); and 4(2), 31–36 (1981); S. Shen, *Sci. Am.*, 104–110 (April 1987); T. D. Rossing *et al.*, *J. Acoust. Soc. Am.* 81, 369–373 (1988)] from the Shang (ca. 1500–1050 BC) and Zhou (ca. 1050–256 BC) dynasties were compared, and the patterns of pitch distribution from octave to octave were scrutinized. Over time, an increasing mastery of the technology of manufacturing two-pitch bells can be observed, especially regarding the ability of manipulating the two fundamentals on each bell in such a way that both could be used in music making. From Western Zhou (ca. 1050–771 BC) onward, the pitch distributions of each chime show a basic division into a lower and a higher register, the former at first manifesting a greater density of pitches than the latter. In the higher register, the Western Zhou four-tone (mi-sol/la-do) pattern was later gradually enriched with additional pitches. The ever denser distribution of pitches seems to reflect the development of bell chimes from a musical instrument of merely accompanatory nature to one on which melodies could be played in full, sometimes in more than one tonality. Caution must be used, however, in generalizing from these phenomena on the development of tonal scales in China.

2:30**M3. Modal studies in a double second Caribbean steel drum.** Uwe J. Hansen (Department of Physics, Indiana State University, Terre Haute, IN 47803), Thomas D. Rossing, and Scott Hampton (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

Caribbean steel drums characteristically contain several sections of separately tuned areas imbedded in one quasihemispherically shaped drum head that is formed in the end of a 55-gallon steel drum. The double second pan under study for this report includes three A^b sections tuned in octaves. Pitch and spectral content for each section depend on section geometry, mass distribution, and elastic properties. Section geometry is determined by hammer-embossed boundary delineation while mass and tension are modified by selective hammering in the section interior, which enables tuning of several lower partials harmonically. Spectral nearfield studies illustrate harmonic content as well as departures from harmonicity in some of the notes under study. Time average holographic interferometry shows strong amplitude dependence of spectral content of notes and illustrates coupling between sections. Complications in using impact excited modal analysis due to noncoherent input contributions are discussed, and computer-animated modes are illustrated.

M4. Vibrational modes of drumheads in various types of drums. Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

As vibrating systems, drums can be divided into three categories: those consisting of a single membrane coupled to an enclosed air cavity (e.g., kettledrums); those consisting of a single membrane open to the air on both sides (e.g., tom toms, congas); and those consisting of two membranes coupled by an enclosed air cavity (e.g., bass drums, snare drums). In kettledrums, the enclosed air raises the frequencies of the axisymmetric modes while air loading lowers the frequencies of the musically important modes with one to four nodal diameters, and shifts them into a nearly harmonic series. In two-headed drums, coupling of the two heads through the air cavity results in mode pairs in which the heads move either parallel or antiparallel to each other. Dipole, quadrupole, and other multipole symmetries are observed in the radiated sound fields. Some drums, such as the kettledrum and tabla, convey a strong sense of pitch; others such as the bass drum and snare drum do not. The sense of pitch may be enhanced by loading a portion of the drumhead, as in the Indian tabla and mridanga.

3:30

M5. Calculations of timpani normal modes for arbitrary kettle shapes. Robert E. Davis and Arnold Tubis (Department of Physics, Purdue University, West Lafayette, IN 47907)

A practical boundary-integral-equation method is developed for extending a previous study of timpani normal modes [R. S. Christian *et al.*, *J. Acoust. Soc. Am.* **76**, 1336–1345 (1984)] to the case of rigid kettles that are azimuthally symmetric but have arbitrary contours. Room acoustics effects are ignored. Calculations and comparisons of results with experimental data are carried out for various kettle shapes including cylindrical, hemispherical, hemispherical with a cylindrical sleeve, hemiellipsoidal, and paraboloidal. It is found that the closeness of the musically important modal frequency ratios, $f_{11}:f_{21}:f_{31}:f_{41}$ to 2:3:4:5, can be quantitatively accounted for by calculations of the effect of air loading. These desirable ratios may be approximately achieved if $V/\pi a^3 \gtrsim 1.08$ and $\rho_0 a/\sigma \approx 1.51$, where V is the kettle volume, a is the timpani membrane radius, ρ_0 is the mass density of air, and σ is a real mass density of the membrane. The timpani spectrum depends weakly on the detailed kettle shape as long as these conditions are satisfied. An infinite rigid baffle in the membrane plane may be used to greatly simplify the calculations without having a significant effect on the results.

Contributed Papers

4:00

M6. Measuring the characteristic properties of mallets. Ingolf Bork (Physikalisch-Technische Bundesanstalt, D-3300, Braunschweig, West Germany)

A measurement procedure is described that makes it possible to characterize different types of mallets for percussion instruments. From spectral analysis of the impulsive force generated by the mallet head, a frequency range is obtained over which each mallet optimally excites vibrations. The dependence of this optimum range on the strength of the blow constitutes a quality criterion that is also suitable for characterizing piano hammers.

4:15

M7. Sound spectra of ancient Chinese bells in the Shanghai Museum. Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115) and Zhu Hong-Fan (Shanghai Museum, 16 South Henan Road, Shanghai, People's Republic of China)

Sound spectra were obtained from ten bells from the 5th–12th centuries BC by striking them at the appropriate *sui* and *gu* locations. Although considerably different in size, shape, and ornamentation, all the bells had oval cross sections and thus exhibited two-tone behavior [T. D. Rossing *et al.*, *J. Acoust. Soc. Am.* **83**, 369 (1988)]. Frequency ratios of the two fundamental (2,0) modes range from 1.11–1.31 (181–473 ϕ). In most spectra, it is possible to identify two partials most likely associated with

two (3,0) modes that have frequencies 2.32–3.29 times the lowest (2,0) mode. The spectra of these bells are compared to those of other Chinese bells, including those of the renowned Zenghou Yi.

4:30

M8. Modes of vibration and sound radiation from a snare drum. Zhao Huan and Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

The orchestral snare drum is a two-headed instrument about 35 cm in diameter and 13–20 cm deep. Snares of metal or gut are stretched across the lower (snare) head, which is generally thinner than the upper (batter) head. The drum can be played with or without the snares. Coupling of the two heads through the air enclosed by the shell results in mode pairs in which the heads move parallel or antiparallel to each other. When struck near the center, the sound spectrum is dominated initially by a strong partial (typically around 180–200 Hz) due to parallel (01) motion of the two heads. This mode radiates energy rapidly, so after a short time, its amplitude falls below that of the modes with one or more nodal diameters. Partial(s) due to the latter modes dominate the spectrum when the drum is struck off-center. Dipole, quadrupole, and other multipole symmetries are observed in the radiated sound field. Vibrations of the shell contribute very little to the sound spectrum. [Research supported by Remo, Incorporated.]

Session N. Physiological Acoustics III and Psychological Acoustics III: Interactions Between Neurophysiology and Psychoacoustics and Others (Poster Session)

Evan M. Relkin, Chairman

Institute for Sensory Research, Syracuse University, Merrill Lane, Syracuse, New York 13244

Contributed Papers

All posters will be on display from 1:30 to 5:00 p.m. (Viewing hours will also be extended into the evening.) To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 to 4:30 p.m. Contributors are encouraged to leave their posters in place until 9:45 p.m. A cash bar will be set up in the evening to facilitate informal discussion, and the Goldstein Auditorium will be closed at 10:00 p.m.

N1. Physiological "release from masking." J. B. Mott, L. P. McDonald, and D. G. Sinex (University of California, Los Alamos National Laboratory, Life Sciences Division, LS-1 M882, Los Alamos, NM 87545)

Psychophysical "release from masking" [S. Buus, *J. Acoust. Soc. Am.* **78**, 1958-1965 (1985)] is observed when the masking exerted by stimuli with equal power but different envelopes is compared. Depending on the masker-to-probe frequency ratio, thresholds for flat-envelope maskers are higher than those for corresponding fluctuating-envelope maskers. A similar phenomenon was previously reported for masked thresholds based on rate responses of chinchilla auditory-nerve fibers [J. B. Mott *et al.*, *Absts. A.R.O.* **12** (1989)]. Those results were interpreted as suggesting that neural activity during periods of low masker energy was important for detection. The present study examined whether synchronized responses of auditory-nerve fibers also contribute to "release from masking." Synchronized rates were measured in short-time windows, centered on periods of high and low energy in the maskers' envelopes. Synchronized rate at the probe frequency grew more rapidly as a function of level during low-energy windows than high-energy windows. Statistical thresholds based on synchronized behavior will also be reported. [Work supported by NINCDS NS23242.]

N2. Time-domain analysis of auditory-nerve fiber firing rates. Hugh E. Secker-Walker and Campbell L. Searle (Room 20C-014, MIT, Cambridge, MA 02139)

A time-domain analysis of firing-rate data from over 200 fibers from the auditory nerve of cat has been used to estimate the formants of synthetic speech syllables. Distinct groups of fibers are apparent in the neurograms of the firing-rate responses. The intervals between peaks in the firing-rates of the fibers in each group are very similar, and reflect the period of the formant that dominates the group's response. Analysis of these intervals confirms that they correspond directly to the formant periods. It is concluded that the cochlear filters have much shorter impulse responses than the formants to which they respond. The time-domain analysis tracks the changes of lower frequency formants with more precision than previous analyses of the same neural data [M. I. Miller and M. B. Sachs, *J. Acoust. Soc. Am.* **74**, 502-517 (1983); S. A. Shamma, *J. Acoust. Soc. Am.* **78**, 1622-1632 (1985)]. The direct representation of the formant period in the time domain is contrasted with the diffuse spectral representation of the formant, the dependence of spectral peaks on nonformant parameters, and the distortion of the spectrum by physiological nonlinearities.

N3. Frequency and duration discrimination by patients with central auditory pathway lesions. Mary Ellen Thompson and Sharon M. Abel (Mount Sinai Hospital Research Institute, Suite 843, 600 University Avenue, Toronto, Ontario M5G 1X5, Canada)

The performance of three groups of patients with central auditory pathology including subjects with lesions localized to the right temporal lobe, left temporal lobe, and eighth nerve was compared using three psychophysical tasks. These included detection, frequency discrimination, and duration discrimination. Speech perception was measured using the Four Alternative Auditory Feature Test. The patients were compared to a group of age-matched normal control subjects and hospitalized control subjects. Except for patients with acoustic neuromas, all subjects had normal hearing acuity. Performance deficits relative to normal were dependent on site of brain lesion, verified by CT and/or MRI scans. The relationship between the processing of frequency and duration and speech perception will be discussed. [Work supported by the Medical Research Council and National Sciences and Engineering Research Council.]

N4. Auditory-nerve representation of communication signals in background noise. Joshua J. Schwartz, Michael Ferragamo, and Andrea Megala Simmons (Department of Psychology, Brown University, Providence, RI 02912)

Anuran auditory-nerve fibers phase-lock to the periodicity of complex acoustic signals, even though many of these fibers do not synchronize to pure tones at their best frequencies (BFs). In this experiment, the effects of background masking noise on peripheral coding of both spectral and temporal structure of the male bullfrog's species-specific advertisement call was examined. Synthetic advertisement calls with 21 components (periodicity of 100 Hz) were digitally constructed and presented via a closed-field system against different levels of noise. Plots of average discharge rate versus BF show peaks corresponding to the periodicity and low-frequency spectral energy in the signal when presented at noise levels near threshold. These profiles change substantially in the presence of high levels of background noise. Plots of average localized synchronized rate (ALSR) against BF are much less influenced by noise. The data suggest that temporal coding of this natural call is more resistant to noise masking than is a representation based solely on discharge rate. [Work supported by NIH.]

N5. Relations between neural-count and intensity-jnd functions for single auditory-nerve fibers. William S. Hellman (Department of Physics, Boston University, Boston, MA 02215) and Rhona P. Hellman (Department of Psychology, Northeastern University, Boston, MA 02115)

The equation $d'\sigma(x) = \Delta N(x)$, where $x = I/I_0$, is applied to physiological measurements reported by Delgutte [*The Psychophysics of Speech Perception* (1987)] for single auditory-nerve fibers. Intensity-jnd functions are derived for individual fibers with the neural-count function as input. The experimentally obtained neural counts are represented by a hyperbolic tangent function, and the measured dependence of the count variance on the count is approximated by a combination of linear functions. Here, $\Delta N(x)$ can be obtained without approximation for arbitrary values of Δx for a hyperbolic tangent function; it can also be represented by a Taylor series in Δx . The exact equation for $\Delta N(x)$ gives a U-shaped jnd function in agreement with the numerical procedure used by Delgutte. By comparison, the Taylor series for $\Delta N(x)$ converges to the exact solution in fifth order. [Partially supported by the Rehabilitation Research and Development Service of the VA.]

N6. Simulating pitch perception and sensitivity to phase using a peripheral auditory model. Ray Meddis and Michael J. Hewitt (Department of Human Sciences, University of Technology, Loughborough, LE11 3TU, United Kingdom)

The ability of neurophysiological studies to give an adequate account of human pitch perception was studied using a peripheral auditory model comprising (a) outer-ear, low-frequency attenuation, (b) a set of 60 digital filters to simulate basilar membrane frequency selectivity, (c) an array of inner hair cell simulators, and (d) a system based on Licklider's [*Experientia* 7, 128-133 (1951)] theory of pitch perception that generates running histograms of the time intervals among all spikes in fibers originating from adjacent sites on the membrane. The most frequently occurring time intervals were taken to represent the period of the perceived pitch, while the overall pattern of time intervals represented the perceived timbre. The model was tested, giving satisfactory results, using a wide range of harmonic, inharmonic, and noise stimuli known to give rise to pitch perceptions as well as stimuli, where the relative phase of the harmonic components affects stimulus timbre. [Work supported by SERC.]

N7. Alternate-state otoacoustic emissions as a basis for episodic intermittent tinnitus. Edward M. Burns (University of Washington, JG-15, Seattle, WA 98195)

The correlational evidence is so overwhelming that the vast majority of occurrences of severe, continuously present, tinnitus are not associated with spontaneous otoacoustic emissions (SOAEs). Similarly, subjects with SOAEs are rarely subjectively aware of their presence, a fact that is not incompatible with the literature on the adaptation of low-level sinusoidal stimuli. However, OAE researchers have often speculated that unstable SOAEs could be the basis of intermittent tinnitus. This paper reports on measurements made on one subject during an episode of intermittent tinnitus (reported by the subject to be audible in normal circumstances, loud, and extremely annoying), which demonstrate fairly conclusively that the tinnitus in this case is associated with alternate-state SOAEs.

N8. Spontaneous otoacoustic emissions in *Rana catesbiana*, the American bullfrog. T. J. Gennosa (Department of Biomedical Engineering, Boston University, 44 Cummings Street, Boston, MA 02215), H. F. Voigt (Departments of Biomedical Engineering and Otolaryngology, Boston University, Boston, MA 02215), and R. Burkard (Department of Communication Disorders, Boston University, Boston, MA 02215)

Temperature-dependent spontaneous otoacoustic emissions (SOAEs) are reported for the first time in *Rana catesbiana*, the American

bullfrog. A low-noise acoustic measurement system was developed for recording SOAEs in a readily available species of frog. SOAEs were detected in one of five anesthetized frogs during late autumn. The frequency of the emission was temperature dependent, increasing from 1612-1762 Hz, with a change in body temperature from 22.1°-25.3°C. The SOAE level ranged from -3 to +4 dB SPL and was uncorrelated with temperature over the range studied. SOAEs have been reported in *Rana esculenta* [A. R. Palmer and J. P. Wilson, *J. Physiol.* 324, 66P (1981)] and *Rana temporaria* [Wilson *et al.*, in *Auditory Frequency Selectivity*, edited by B. C. J. Moore and R. D. Patterson (Plenum, New York, 1986), pp. 39-46]. The emissions in these other species are also temperature dependent [P. van Dijk and H. P. Wit, *J. Acoust. Soc. Am.* 82, 2147-2150 (1987)]. Experiments are recommended to explore the possibility that the presence and incidence of SOAEs are seasonally dependent in this species. [Work supported in part by NIH.]

N9. Scattered ambient noise as an auditory stimulus for fish. Peter H. Rogers, Thomas N. Lewis, Martha J. Willis, and Scott Abrahamson (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

In an earlier paper [P. H. Rogers, *J. Acoust. Soc. Am. Suppl.* 1 79, S22 (1986)], it was hypothesized that one role of the fish's auditory system may be to detect and localize nearby fish by "imaging" ambient noise scattered by their swim bladders. This is analogous to the role of the visual system of most animals, where the relevant signal is ambient light scattered by objects rather than light emitted by luminous objects. A classical conditioning experiment has been performed which indicates that the fish auditory system is capable of functioning in this manner. The ambient noise is provided by a J-9 transducer driven by Gaussian noise. Scattering of this noise by the resonant swim bladder is simulated by applying a filtered version of the noise signal to a small spherical projector. The "target strength" is bandwidth of the filter. The fish is conditioned to respond to the presence of the signal from the spherical projector. The fish's ability to detect this signal (as a function of range, bearing, etc.) is taken to be a measure of its ability to detect scattered ambient noise.

N10. The amplitude-latency trade-off demonstrates a temporal basis for fine echo delay acuity in bat sonar. J. A. Simmons, M. Ferragamo, and C. F. Moss (Department of Psychology and Section of Neurobiology, Division of Biology and Medicine, Brown University, Providence, RI 02912)

Echolocating bats (*Eptesicus fuscus*) can perceive a change in echo delay smaller than 1 μ s in two-choice, echo-jitter discrimination tasks. Although the bats emit sounds short enough to avoid overlap of stimulus echoes with incidental echoes, and although the transfer function of the delay electronics reveals no amplitude or spectral cues correlated with delay, it is difficult to believe that such fine temporal is not based on spectral cues. If the delay of echoes is encoded by neural discharge timing rather than from spectral information, perceived delay should shift according to the neural amplitude-latency trading function, which is -13 to -18 μ s/dB for N_1 - N_4 responses in *Eptesicus*. The apparent delay of echoes in the jitter task shifts by -16 to -17 μ s/dB, demonstrating that the bats perceive the delay of jittered echoes from their arrival time rather than from some spectral consequence of the particular value of echo delay. The corresponding shift of the entire jitter psychometric function demonstrates that bats may perceive the phase of echoes relative to emissions from information encoded in the timing of neural discharges, as previously proposed from coincidence of the jitter psychometric function with the cross-correlation function of echoes.

N11. Perilymph volume flow in the fistulized cochlea. Naoki Inamura, Alec N. Salt, Ruediger Thalmann, and Arti Vora (Department of Otolaryngology, Washington University, St. Louis, MO 63110)

It has been previously reported that in guinea pigs the rate of perilymph volume flow is extremely low (< 2 nl/min) when the cochlea is in the physiological, sealed state. In the present study, flow rates were monitored when the otic capsule was perforated. Flow was measured by comparing dispersal of a tracer (trimethylphenylammonium) in both directions along scala tympani with ion selective microelectrodes. Perforation of the cochlear apex produced a high rate of flow averaging $0.4 \mu\text{l/min}$. The rate was reduced considerably when cerebrospinal fluid (CSF) pressure was released by opening the dura at the foramen magnum. Under these conditions, the mean rate of 6.9 nl/min. Blockage of the cochlear aqueduct was even more effective, reducing flow to a mean value of 1.4 nl/min. In the latter condition, the flow rate was not significantly different from the normal, sealed cochlea condition. These results demonstrate that the high rates of flow in the fistulized cochlea arise from a nonphysiologic entry of CSF into scala tympani through the cochlear aqueduct. For experimental purposes, this nonphysiologic flow can be reduced by releasing CSF pressure, or better still, by occluding the cochlear aqueduct. [Work supported by NIH Program Project Grant No. PO1 NS24372.]

N12. A wide dynamic range speech test for the evaluation of compression hearing aids. Arlene C. Neuman, Matthew H. Bakke, and Harry Levitt (CUNY Graduate School, 33 West 42 Street, New York, NY 10036)

Wide dynamic range test materials have been developed for the evaluation of compression hearing aids. Anomalous sentences containing four monosyllabic test words per sentence were recorded by a female talker. The four test words in each sentence were then scaled (using a waveform editing system) to represent four different sound-pressure levels covering a 30-dB range (in 10-dB steps). Performance on this speech material was compared to a performance-intensity function obtained using monosyllabic word lists at the same four sound-pressure levels. Testing was done for a linear and two compression systems (compression ratio of 2.5, instantaneous attack time, release times of 20 and 140 ms). For all three conditions, frequency shaping based on a half-gain rule was used. Ten listeners with sensorineural hearing loss served as subjects. Results of the experiment indicated that the wide dynamic range test material was sensitive to the dynamics of the compression system being evaluated, while the monosyllabic word materials presented at discrete levels was not. [Work supported by NIH Grant No. PO1 NS17764.]

N13. Comparison of long-term average speech spectrum at ear-level and standard recording positions. Leonard E. Cornelisse, Jean-Pierre Gagné, and Richard C. Seewald (Department of Communicative Disorders, University of Western Ontario, London, Ontario N6G 1H1, Canada)

The long-term average speech spectrum (LTASS) was obtained from ten male and ten female talkers for two microphone positions. Dual channel recordings of speech were recorded and played back for spectrum analysis. One channel recorded the speech signal obtained from a microphone placed 30 cm directly in front of the talker, while the second channel recorded the speech signal obtained from a microphone placed at the talker's ear. A spectrum analyzer (BK type 4132) was used to compute the overall levels and 1/3-octave-band levels. The analyses indicated significant differences in overall and the 1/3-octave-band levels between the two recording positions. The mean overall level at the position of the ear was approximately 6 dB higher than the overall level at the standard recording position. Also, 1/3-octave-band analyses indicated that the ear-level LTASS consisted of more low-frequency energy and less high-frequency energy than the LTASS recorded at the standard position. The implications of these results for hearing aid selection will be addressed.

N14. The P300 response to verbally presented digits. Lawrence F. Molt (Department of Exceptional Student Education, Florida Atlantic University, Boca Raton, FL 33431) and Richard S. Saul (Audiology and Speech Pathology Service, Veterans Administration Medical Center, Miami, FL 33125)

Latency and amplitude functions for the P300 component of the slow auditory evoked potential were measured for 20 normal hearing adults. Stimuli consisted of digitized numbers, 1-10 (excluding 7), of 480-ms duration. The method of presentation for stimuli pairs was in an oddball paradigm in which the frequently occurring digit was presented 80% of the time and the rarely occurring digit 20% of the time. Pairs consisted of acoustically similar or acoustically divergent waveforms (e.g., 4-5 vs 8-2, respectively). Evoked potentials were recorded from F_z , C_z , and P_z sites. While both mean latency and mean amplitude values differed among the three sites, with the greatest amplitude and shortest latency recorded from site C_z , differences were not statistically significant. Similarly, mean amplitude and latency value differences did not reach statistical significance for acoustically similar versus acoustically divergent digit pairs at any of the three sites.

N15. Effects of intense sound exposures and aspirin on overshoot. Craig A. Champlin and Dennis McFadden (Departments of Psychology and Speech Communication and Institute for Neurological Sciences Research, University of Texas, Austin, TX 78712)

Brief tonal signals were presented 2 or 190 ms following the onset of a 200-ms broadband masker. These two conditions of signal delay were tested before and after a series of exposures to a tone intense enough to induce temporary threshold shifts (TTS). The difference in detectability between the short- and long-delay signals (i.e., overshoot) was reduced when a TTS of at least 10 dB was induced, but not when smaller amounts of TTS were induced. This reduction in overshoot was due to an improvement in the postexposure detectability of the signal delayed by 2 ms; the detectability of the signal delayed by 190 ms was unchanged pre- and postexposure. Recovery from both the TTS and the reduction of the overshoot followed slightly different time courses. Data are currently being collected for the two conditions of signal delay before and after the production of an aspirin-induced threshold shift. [Work supported by NIH.]

N16. Comodulation masking release for three types of modulator as a function of modulation rate. Robert P. Carlyon, Søren Buus, and Mary Florentine (Communication Research Laboratory (226 FR), Northeastern University, Boston, MA 02115)

To investigate the dependence of comodulation masking release (CMR) on the type and frequency of modulator, thresholds were measured for a 4-kHz tone masked by modulated and unmodulated noises. The maskers were a 400-Hz wideband of noise centered on 4 kHz (NOCUE), the same noise with a 2700-Hz low-pass noise added (LPCUE), and a wideband noise with a passband between 3 and 6 kHz (WBCUE). Each modulated masker was presented with several modulation frequencies, f_m s and three types of modulators: a square wave, a sine wave, and a low-pass noise. Thresholds were lower in the presence than in the absence of modulation for all modulator types, except when f_c was high. The modulated-unmodulated difference (MUD) decreased monotonically with increasing f_c in all conditions. For any given f_c , MUDs were greater for square-wave and low-pass-noise modulation than for sinusoidal modulation. The CMR, defined as the difference between the MUD obtained in the NOCUE condition and that obtained in each cue condition, did not vary systematically with f_c over the range studied. CMR was smaller for sinusoidal modulation than for either square-wave or low-pass-noise modulation. Although there was some variability across listeners, low-pass-noise modulators that produced an f_c of between 32 and 100 Hz resulted in substantial (5-9 dB) CMRs for all three listeners. [Work supported by the Piggott-Wernher Fellowship and NIH.]

N17. Masking patterns for comodulated and uncomodulated narrow-band noises as a function of masker band separation. Rebecca M. Fischer and D. Wesley Grantham (Division of Hearing and Speech Sciences, Vanderbilt University, 1114 19th Avenue South, Nashville, TN 37212)

Masking patterns were determined for a pair of narrow-band (25 Hz wide) noises whose envelopes were perfectly correlated, uncorrelated, or negatively correlated. In one condition, noise bands were centered at 962.5 and 1037.5 Hz in order to examine the effects of comodulation masking release (CMR) when paired noise bands were within one critical band; the second condition used noise bands centered at 887.5 and 1112.5 Hz, a separation of greater than one critical bandwidth. Signal thresholds were determined for a range of frequencies (750–1250 Hz) above and below those of the maskers in order to examine the process of CMR outside the immediate spectral region of the maskers. Preliminary data suggest that CMR gradually decreases as the signal is moved away in frequency from the center of either noise band. A comparison of CMR in positively and negatively comodulated conditions is expected to shed light on the type of mechanisms underlying CMR within and across critical bands.

N18. Profile analysis by parallel distributed networks. James H. Howard, Jr., Jeffrey L. Harpster, and Maxwell H. Miller (Department of Psychology, The Catholic University, Washington, DC 20064)

Several profile analysis (spectral shape discrimination) phenomena were simulated using simple, three-layer, feed-forward, parallel networks. As in previous studies with human listeners, pairs of multitonal sounds were used—one with all components at a uniform, background level and the other with a tonal signal added in-phase to the central component. Networks were trained to select the signal alternative in a 2AFC paradigm using the generalized delta rule with error propagation and were then tested under various conditions. Background levels were varied randomly either between or within trials. In the former case, the network learned to compare energy at the signal frequency across the two sounds (intensity discrimination), whereas, in the latter case, it learned to compare the distribution of spectral energy within sounds (profile analysis). These results mimic previous findings with human listeners reported by Green and his colleagues [D. M. Green, *Profile Analysis* (Oxford U. P., New York, 1988)]. In follow-up simulations, improved profile analysis have been demonstrated when the frequency range of tonal components is expanded and as the duration of the complex increases to 100 ms. [Work supported by ONR.]

N19. Spectral pattern discrimination in children. Prudence Allen and Frederic Wightman (Waisman Center, University of Wisconsin, Madison, WI 53705-2280)

This experiment used a video-game-based adaptive forced-choice paradigm to assess the effect of noise on the ability of children to discriminate among stimuli with different spectral shape. The stimuli were synthesized vowels and consonants, and tonal complexes that had sinusoidally rippled spectra. Preschool children, school-aged children, and adults served as subjects. In order to encourage the subjects to base their discriminations on spectral shape rather than on local energy concentrations, the overall level of the stimuli was randomized over a 20-dB range for each presentation. In general, the results suggest that spectral pattern discrimination ability for all stimulus types improves with age. The age at which the children's performance reaches adult levels varies with stimulus type. Moreover, noise has a particularly deleterious effect on the spectral pattern discrimination ability of the younger subjects. [Work supported by DRF and the Waisman Center Rowan Trust.]

N20. Perception of pitch structure in pure-tone melodic sequences. Nicholas Oram and Lola L. Cuddy (Department of Psychology, Queen's University, Kingston, Ontario K7L 3N6, Canada)

Musically naive subjects were tested in two experiments investigating the perception of pitch structure in pure-tone sequences. Each sequence contained 20 successive samples from either a diatonic or a nondiatonic toneset of seven tones with probability of occurrence of each tone systematically varied. Sequences were presented in either of two frequency ranges (262–494 Hz, or 1047–1976 Hz). Frequency range was varied between subjects in experiment 1 and within subjects in experiment 2. On each trial, a sequence was followed by a probe tone, one of the 12 chromatic tones within the same octave as the sequence. Subjects rated the goodness-of-fit of the probe tone to the sequence. In both experiments, probe-tone ratings were positively related to probability of occurrence of the probe tone in the sequence. Moreover, for sequences generated from diatonic tonesets and presented in the high-frequency range, ratings also reflected the tonal hierarchy implicated by the key of the toneset. [Work supported by NSERC.]

N21. Reaction time measures of phantom image recognition in stereophonic listening. H. Brazzil (Department of Mathematics, University of St. Thomas, Houston, TX 77006) and A. Yonovitz (Speech and Hearing Institute, University of Texas Health Science Center at Houston, Houston, TX 77030)

Intensity stereophony allows the listener to perceive phantom images, which may be localized at any point on the imaginary soundstage existing between loudspeaker pairs. Specification of the phantom image typically relies on subjective evaluation by the listener. This study utilized reaction time to quantify the ability of listeners to accurately identify the image position as a function of frequency. Subjects were seated at 30-deg incidence to each loudspeaker in an anechoic environment. Each trial consisted of a pair of third-octave noise bursts at one of six frequencies (250–8000 Hz). The first of each stimulus pair was presented with no interloudspeaker intensity difference (phantom center). The second stimulus occurred randomly at phantom center, phantom left, or phantom right, and subjects were instructed to correctly respond as quickly as possible in this three-choice forced response task. Interloudspeaker intensity difference for images occurring left or right of center was 2, 4, or 6 dB. Longer reaction times occurred for center image responses, independent of frequency. Shortest reaction times were obtained at 2 kHz, with increasing latencies occurring at higher frequencies. Data suggests that phantom image identification in stereophonic listening is compromised above 2 kHz.

N22. Binaural signal detectability in noise amplitude-modulated (NAM) noise as a function of masker bandwidth. Mark Hedrick and D. Wesley Grantham (Bill Wilkerson Hearing and Speech Center, 1114 19th Avenue South, Nashville, TN 37212)

Thresholds were determined for a 500-Hz tonal signal in the presence of a noise amplitude-modulated (NAM) noise, where the carrier was a wideband Gaussian noise and the modulator was a 20-Hz-wide noise band centered at 20 Hz. The NAM noise was filtered after modulation to produce a band centered on the 500-Hz signal. Both diotic (So) and interaurally phase-reversed ($S\pi$) signal thresholds were determined as a function of masker bandwidth (10–1000 Hz). When the masker was presented diotically, So thresholds (and to a lesser extent, $S\pi$ thresholds) first increased, then decreased as a function of masker bandwidth, thus showing the effects of comodulation masking release at the wider bandwidths. The masking-level difference (MLD, i.e., the difference between So and $S\pi$ thresholds) increased as masker bandwidth decreased, as it did with a random (unmodulated) noise masker. When the masker was presented such that the noise carrier was diotic but the noise modulators were inde-

pendent at the two ears, a sizable MLD still occurred (5–10 dB), but the MLD decreased with decreasing bandwidth. The decreased MLD at small bandwidths resulted from the higher $S\pi$ thresholds with this stimulus than with a true diotic masker, which can be explained by the slowly fluctuating interaural intensity differences that this type of masker produces at narrow bandwidths. [Work supported by NIH.]

N23. Binaural frequency-following response (FFR) to lateralization stimuli. B. B. Ballachanda and G. Moushegian (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47906 and Callier Center-UTD, Dallas, TX 75235)

The cues for localization or lateralization are interaural disparities of time and intensity, and neurophysiological studies have elegantly demonstrated that lower-brain-stem neurons, in conformity with anatomical observations, are differentially sensitive to the stimulus parameters that mediate binaural phenomena. Brain-stem responses were obtained using low-frequency (500 Hz) stimuli in various combinations of interaural phase (time) and interaural phase and level presented in-concert and in-opposition through earphones. The findings show that the amplitudes and morphologies of the volume-conducted evoked FFR potentials are differentially altered as a function of time and level variables. There are significant reductions, for example, in the amplitude of the evoked responses due to interaural time differences. Furthermore, for in-concert and in-opposition time and level stimuli, the amplitude changes are distinctly different, reflecting the complex neuronal interactions at the level of the brain-stem nuclei and tracts.

N24. Comparisons of between- and within-subject variability in repeated-measured auditory brain-stem responses (ABRs) in 5–6-year-old children. Janiece M. Lord-Maes and Judith L. Lauter (Departments of Educational Psychology and Speech and Hearing Sciences, University of Arizona, Tucson, AZ 85721)

In reports to this Society, and publications [J. L. Lauter and R. L. Loomis, *Scand. Audiol.* **15**, 167–172 (1986); *Scand. Audiol.* **17**, 87–92 (1988)], results of repeated-measures ABR testing in young adults have been described, indicating the variability of peak parameters, such as latency and amplitude, provides information that absolute values of these parameters do not: contrasts in between- versus within-subject consistency, and by ear of stimulation. Seven 5–6-year-old children were tested in eight weekly sessions for ABRs to right, left, and binaural clicks, and see adultlike patterns based on nonadult values: (1) contrast in between- versus within-subject consistency; (2) peak differences; and (3) ear differences. Preliminary comparison between these data, a study on 10-year-old males, and 15 young adults suggest that ABR variability may be sensitive to auditory-system developmental changes that continue long after ABR absolute peak latencies have achieved “adult” values. [Work supported in part by AFOSR.]

N25. Comparison of rate effects on the auditory middle latency evoked response (MLR) in young and elderly adults. Jody Newman Ryan and George Moushegian (Callier Center for Communication Disorders, School of Human Development, University of Texas at Dallas, Dallas, TX 75235)

Middle latency responses (MLR) are rate sensitive in infants but less rate sensitive in adults. It has been suggested that developmental factors at both ends of the age continuum may influence MLR [Jerger *et al.*, *Sem. Hear.* **9**, 75–86 (1988)]. Effects of rate in the elderly are unknown. This study was undertaken to determine how rate alters the waveforms of the MLR in elderly subjects. Tone bursts of 500 and 2000 Hz were presented

at two sensation levels (30 and 60 dB SL) monaurally and binaurally at six repetition rates: 3.1, 9.1, 15.1, 27.1, 39.1 (the 40-Hz, or steady-state, response), and 51.1/s. The responses of young (under 35 years) and older (over 60 years) subjects were compared. The results revealed that the waveforms are similar for the two groups at different rates, with the exception of latency prolongations, especially for the later waves (e.g., Pb at 50–60 ms), in the older subjects. Furthermore, at the higher rates, the amplitudes of the waveforms of the older subjects are larger than those in the younger group.

N26. Binaural difference in rat. Maria Psaltikidou and Roger P. Gaumond (Bioengineering Program, Pennsylvania State University, University Park, PA 16802)

The binaural difference (BD) is the sum of monaural acoustic brain-stem responses (ABRs) minus the binaural ABR. Its construction implies operation of a nonlinear element with binaural input. The BD in five rats was studied using insert earphones that provided 60 dB of interaural isolation measured in the occluded external meatus. Scalp to chin BD consisted of a double-peaked complex with a following trough occurring during peaks IV and V of the sum of monaural ABRs. BD magnitude was defined as the maximum peak-to-trough waveform amplitude. As stimulus intensity increased, BD magnitude increased linearly with the magnitude of ABR peak I. A bilaterally symmetric stimulus generally yielded smaller BD magnitude than did stimuli with small bilateral asymmetry (8-dB interaural intensity difference, 64- μ s interaural time difference). With larger asymmetry, BD magnitude decreased. The linear increase of BD magnitude with ABR peak I magnitude is consistent with operation of an element that applies a rectification nonlinearity to the difference of binaural signals during peaks IV and V of the ABR.

N27. Effects of homophasic and antiphase stimulation on the binaurally evoked auditory brain-stem responses. Vishakha W. Rawool and Bopanna B. Ballachanda (Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907)

This study investigated the effects of homophasic and antiphase stimulation on the binaurally evoked auditory brain-stem responses (ABRs). Ten normal subjects were presented with two in-phase (binaural condensation and binaural rarefaction) and two out-of-phase (left ear condensation, right ear rarefaction and right ear condensation, left ear rarefaction) stimulus configurations. The stimuli were 100- μ s clicks delivered at 47 dB SL (repetition rate 5.6/s). The ABRs were picked up between high forehead (active) and left earlobe (reference), the nasion served as ground. The earlier of the latencies obtained with homophasic stimulation was compared with the earlier of the latencies obtained with antiphase stimulation. A subject-by-treatment analyses on the peaks (P) and troughs (N) of each component revealed significantly earlier latencies for the antiphase condition for components IIN ($p < 0.05$), IIIP ($p < 0.025$), VIP ($p < 0.01$) and VIN ($p < 0.025$). Similar trends were apparent in components IVP ($p < 0.1$), IVN ($p < 0.2$), and VP ($p < 0.1$). No differences were observed for components IP, IN, IIP, and VN.

N28. Human auditory localization performance in azimuth. Richard L. McKinley and Mark A. Ericson (AAMRL/BBA, Wright-Patterson AFB, OH 45433-0653)

Human auditory localization performance in the free-field, over headphones via a simulator and over headphones via a real-time, head coupled,

digital auditory localization cue synthesizer will be described in the presentation. In the headphone conditions, the head position is monitored to update the sound source position relative to the listener's orientation. The synthesizer presents sounds over headphones that are perceived to be outside of the listener's head in the horizontal plane. The performance experiments were designed using a within subject, repeated measure

study. The stimuli used were wideband pink noise, male speech bandlimited to 3.5 kHz, female speech bandlimited to 6 kHz and octave band noise from 125 Hz to 8 kHz. The mean magnitude error and mean response time were measured at 72 random points for each stimulus. Five male and five female subjects participated. A 4.8-deg mean magnitude error and a 4.4-s mean response time were collected with the synthesizer.

TUESDAY AFTERNOON, 23 MAY 1989

REGENCY B, 1:30 TO 4:40 P.M.

Session O. Structural Acoustics and Vibration II: Modal Analysis

S. P. Ying, Chairman

Gilbert Services, P.O. Box 1498, Reading, Pennsylvania 19603

Chairman's Introduction—1:30

Invited Papers

1:35

O1. Modal analysis technology review. Paul McDonald and Garth Wiley (Structural Dynamics Research Corporation, 2000 Eastman Drive, Milford, OH 45150)

An overview of current experimental modal analysis technology is provided with consideration to its use with finite element analysis. The implications of this available technology to acoustic applications are included. The advantages and disadvantages of various practical modal analysis methods will be covered. A look to the future is provided in assessing the influences of computational and graphics advances anticipated in workstation hardware. Various application examples will be reviewed to illustrate methodologies employed.

2:00

O2. Current and future topics in modal analysis at the University of Cincinnati. Robert W. Rost (Department of Mechanical and Industrial Engineering, University of Cincinnati, Cincinnati, OH 45221-0072)

The University of Cincinnati, Structural Dynamics Research Laboratory, has been involved in the area of modal analysis for over 20 years. Currently, there are 5 faculty members and 15 graduate students who work with UC/SDRL. UC/SDRL has worked with the United States Air Force, NASA, General Motors, and many other industries on a variety of research and consulting topics. Current and future technology in the area of modal analysis will be discussed. Current topics to be discussed will include multi-input/multi-output testing techniques, parameter estimation for multi-input/multi-output testing, and work station environment. Future topics to be discussed will include step sine testing, active control, and new signal processing techniques.

2:25

O3. Modal identification for hysteretic structure. Daniel J. Inman (Department of Mechanical and Aerospace Engineering, University of Buffalo, Buffalo, NY 14260)

This work describes a method of analyzing the vibration response of a viscoelastic beam. The equation of motion of the transverse vibration of a beam with viscous damping and temporal hysteresis is given. Several methods for measuring the constants of a given hysteresis kernel are presented. One is based on a least-squares fit to data using inverse procedures for partial differential equations. The second approach discussed is that of introducing an auxiliary coordinate. This coordinate combined with a transfer function model of the hysteresis effect yields a modal analysis solution for the viscoelastic beam. It is shown that identification is then possible by using modal analysis techniques. This theoretical derivation is compared with test data.

2:50

O4. A new approach to system identification in conjunction with digitally Swept MultiExciter Sine Control. M. A. Underwood and R. C. Stroud (Synergistic Technology, Incorporated, 1333 Lawrence Expressway #410, Santa Clara, CA 95051)

This paper discusses a new adaptive control technique that applies optimization ideas to the general problem of updating an imprecise system Impedance Matrix, I , as part of the feedback control process that is associated with MultiExciter Swept Sinewave Control. Extensions of the approach to other MultiExciter problems such as waveform and random vibration control will also be explored. The effects on the ability of the Adaptive Control Algorithm to converge by low matching of phase and amplitude between the input channels, low coherence between the exciter drive vector and the control point response vector, and low input/output

dynamic range are also analyzed. The effects of the above instrumentation errors on the optimization problem associated with experimental modal analysis are also explored as well as the potential benefits that an enhanced frequency response matrix estimate, as a result of the above system identification process, can provide.

3:15

O5. Modal analysis of rotating machinery. S. P. Ying (Gilbert Services, Inc., P.O. Box 1498, Reading, PA 19603)

Modal analysis of rotating machinery and its supporting structure is discussed. This paper presents the natural frequencies, damping ratios, and mode shapes of various modes of rotating machinery systems such as fans and radar antenna foundations as examples. For a rotating machinery, it is necessary to have a dynamic correction factor for the dynamic natural frequency of a rotor (critical speed). The dynamic correction factor for ventilation fans is presented with examples. The overall natural frequency of an entire system depends on the dynamic combination of the rotor and its supporting structure. If a heavy duty rotor bearing is used, the mode for the rotor having a rigid support becomes significant, which simplifies design, and the dynamic sensitivity of the system is reduced. For a large industrial fan, an early foundation design is compared with an improved foundation design in later stage from a mode shape viewpoint. Based on suggestions developed from modal analyses, a supporting structure was improved, gear noise resulting from torsional vibration was eliminated, and a rotor resonant problem was resolved.

Contributed Papers

3:40

O6. Group delay and reverberation in multi degree of freedom systems. Djamil Boulahbal and Richard H. Lyon (Department of Mechanical Engineering, MIT, Cambridge, MA 02139)

Signals that travel in multi degree of freedom (dof) systems experience group delay that is a function of system geometry, modal density, damping, and signal processing procedures. This paper is interested in developing inverse filters for such systems for the purposes of mechanism and structural diagnostics, and the dereverberation of response signals. The design of inverse filters is greatly aided if the process can begin with approximations to the system transfer functions that are based (at least in part) on conceptual models for the system dynamics. In this paper, the theoretical basis for such models is outlined, including the effects of signal processing procedures. The various elements of these models that have been tested experimentally, and the results of these studies will be presented.

3:55

O7. Resonance characteristics of connected subsystems. Takeru Igusa, Jan D. Achenbach, and Kyung-Won Min (Department of Civil Engineering, Northwestern University, Evanston, IL 60208)

Resonance characteristics of a collection of linear subsystems interconnected by waveguides is examined. The properties of the subsystems are determined separately using modal or mobility analysis, and the topology of the waveguide network is cast in matrix form using a wave propagation approach. It is found that harmonic responses are described by multi-valued analytic functions defined in the complex plane (or a single-valued function on a Riemann surface). The excitation frequency is represented by the unit circle and resonances are poles located outside of the circle. The analytic form of the response functions provide insight into the relationships between the modes of each interconnected subsystem as well as the relationship between modal vibration and wave propagation effects. These relations are complicated by the fact that the interconnecting waveguides by themselves introduce new modes of vibration. The complexity of the problem is reduced by restricting the analysis to a window of complex frequencies, whose width is determined by the degree of interaction between modes of the subsystems and wave guides. The interaction is measured by parameters generalized from an earlier study of simpler subsystems [T. Igusa and A. Der Kiureghian, *J. Eng. Mech.* **101**, 20-41 (1985)]. [Work supported by ONR.]

4:10

O8. Detection of global modes in a foundation and hull structural model from drive point mobility measurements on the foundation. Kevin McCoy and Richard H. Lyon (Departments of Ocean and Mechanical Engineering, MIT, Cambridge, MA 02139)

Closely coupled modes of two connected structures may be considered to be "global modes," involving the combined motion of both structures. Such modes will account for the greatest energy transfer between the structures, and it is therefore desirable to find ways of detecting them. This paper is interested in the case in which it is convenient to make measurements on only one of the two structures, in this case a "foundation model" that is connected to a "hull model." The foundation (a beam) is excited by a shaker, and its drive point mobility is measured. At certain frequencies, an interaction with the hull (plate) modes can be detected. This interaction can be modified by using small masses to change the coupling between the structures and the foundation resonances. In this way, the global modes can be separated truly from structural resonances that simply happen to lie close to each other.

4:25

O9. A generalized approach to random noise synthesis. Keng D. Hsueh and Roger P. Hamernik (Auditory Research Laboratory, 107 Beaumont Hall, State University of New York, Plattsburgh, NY 12901)

A generalized approach to the synthesis of Gaussian and non-Gaussian random noises having a preselected amplitude spectrum has been developed. An application of the inverse discrete Fourier transform following certain manipulations in the frequency domain allows both Gaussian (skewness = 0 and kurtosis = 3) and non-Gaussian (variable skewness, kurtosis, and crest factor) random noises to be constructed. The theoretical background and the results of a numerical simulation will be presented, which demonstrate the functional relation between the various spectral manipulations and the descriptors of the synthesized random noise. Analog signals were synthesized using the above method in conjunction with PC-based instrumentation. The results show that the approach is viable and that the synthesized random waveforms can be easily tailored to simulate a variety of real-world noise/vibration signals, e.g., high kurtosis (impulsive) industrial noises, helicopter noises, and missile vibration signals. The application of such a methodology to acoustical testing, environmental stress screening, structural testing, etc., will be discussed. [Work supported by NIOSH.]

Session P. Underwater Acoustics II: Scattering and Reverberation

Clarence S. Clay, Chairman

Department of Geology and Geophysics, University of Wisconsin, Madison, Wisconsin 53706

Chairman's Introduction—2:00

Contributed Papers

2:05

P1. Acoustic backscatter from an inhomogeneous volume beneath a planar interface: Model and experiment. Paul C. Hines (Defense Research Establishment Atlantic, P. O. Box 1012, Dartmouth, Nova Scotia B2Y 3Z7, Canada)

Mathematical models of acoustic scattering from the ocean bottom currently available, generally fail to predict backscatter levels of sufficient magnitude, for two limiting cases; as the bottom becomes increasingly smooth and as the grazing angle becomes small. A mathematical model of acoustic backscattering is derived herein which attempts to address these shortcomings for the case of a sediment bottom. In the model, scattering results from fluctuations in concentration, where concentration is roughly defined as the water to particle ratio within the sediment. The model allows for penetration of the incident wave into the bottom at subcritical grazing, and retransmission of scattered spherical waves through the (planar) interface. The frequency and grazing angle dependence of the acoustic backscatter are determined primarily by the correlation function of the concentration fluctuations, while the magnitude is controlled by the mean-square value of these fluctuations. To complement the modeling, laboratory experiments were performed to measure volume backscatter from a smooth, water-saturated sand bottom at grazing angles from 10° – 50° for frequencies of 25–200 kHz. Agreement between model and data is good. [Work was performed during post-graduate degree at the University of Bath, Bath, U.K.]

2:20

P2. Acoustic wave scattering by inhomogeneous obstacles in layered media. I. T. Lu and H. K. Jung (Department of Electrical Engineering/Computer Science/Weber Research Institute, Polytechnic University, Farmingdale, NY 11735)

Acoustic wave scattering by homogeneous obstacles in layered media has been studied by a new method that combines the hybrid ray-mode and boundary element methods [I. T. Lu, IMACS Meeting, Princeton Univ., March 1989]. The latter has been employed to formulate the scattering process and the former to provide the Green's function of the layered environment. Here, the new method is extended to analyze wave scattering by inhomogeneous scatterers by including the finite element method. By finite element method, the inhomogeneous obstacles are divided into small elements and the field on these elements are formulated in terms of a system of algebraic equations. By employing the boundary element method, the field distribution along the boundaries of the obstacles are formulated in terms of integral equations which are subsequently reduced to algebraic equations. In the integral equations, one needs to compute the Green's function of the layered environment for various arrangements of locations of source and receiver. The newly developed hybrid ray-mode method is best suited for this purpose because it combines rays and modes self-consistently within a single framework and optimizes the advantage of each. Numerical implementation illustrates these aspects. [Work Supported by NSF.]

2:35

P3. Angle spreading of acoustic energy on bottom bounce paths. Diana F. McCammon (Applied Research Laboratory, The Pennsylvania State University, P. O. Box 30, State College, PA 16804)

A model for the spatial distribution of bottom returning energy has been developed that relies on a combination of surficial roughness and sublayer roughnesses to produce a non-Gaussian probability distribution of returns. Sediment inhomogeneities and sublayering are modeled by the artifice of assuming that the rms slope of the surficial layer increases with grazing angle. Deep region inhomogeneities are modeled by choosing a high value of 22° for the rms slope of the basement. These simplifying assumptions keep the model inbounds computationally and produce the correct behavior for angle spreading in both thick and thin sediment regions. Excellent agreements are shown with data. [Work supported by NAVSEA.]

2:50

P4. Cross correlation of the repeated transmissions of signals scattered at a rough sea floor. C. S. Clay (Department of Geology and Geophysics, University of Wisconsin-Madison, Madison, WI 53706)

Real measurements of the cross correlations of repeated transmissions from a source ship to a receiver on another ship must contend with drifting ships. Over a rough sea floor, the cross correlations depend on the geometry of the experiments, the nature of the roughness, and transmission techniques. Green [J. Acoust. Soc. Am. Suppl. 1 **68**, S72 (1980)] described multiple transmission techniques that can reduce the effects of receiver displacements between transmissions. Since it is easy to control drifting ships in numerical calculations, it was decided to use numerical simulations. The wedge assemblage and facet ensemble method were used to compute impulsive signals scattered at a rough bottom [C. S. Clay and W. A. Kinney, J. Acoust. Soc. Am. **83**, 2126–2133 (1988)]. The results show that one can expect cross correlations as high as 80%–90% for receiver displacements of 25 m between transmissions. [Research supported by Office of Naval Research and the Geophysical and Polar Research Center of the University of Wisconsin-Madison.]

3:05

P5. Computer simulation model useable for testing underwater communication techniques. Lee E. Estes, Gilbert Fain, and Paul R. Caron (Department of ECE, Southeastern Massachusetts University, North Dartmouth, MA 02747)

A model, and its software implementation, that is designed to simulate the transmitted and ocean-degraded received signal of an acoustic underwater transmission is presented. The approach taken for the model is the development of a stochastic impulse response characterized by its rever-

beration statistics and decay time and shape. A correlation time for the evolution of the nonstationary character of the impulse response is also defined. Signal transmission and reception is then modeled by processing signals through two statistically independent realizations of the impulse response. The two realizations are stationary during a correlation interval and are linearly summed after weighing with leap-frogged raised cosines. New realizations are generated for each correlation interval. Results appropriate to transmission in a shallow-water long horizontal channel are presented and compared with experimental results. [Work supported by the Massachusetts Centers of Excellence Corporation.]

3:20

P6. A normal-mode approach to simulation of very low-frequency reverberation in an Arctic environment. Thomas J. Hayward (Naval Research Laboratory, Code 5123, Washington, DC 20375)

A normal-mode-based approach to simulation of very low-frequency underice reverberation in an Arctic environment is presented. Acoustic backscattering from the ice layer and the ocean bottom is simulated using a normal-mode-based model, with underice and bottom backscattering represented by coupling of forward- and backward-traveling waves associated with the normal modes. Results of simulations performed to test the feasibility of associated methods for estimating backscattering strengths from reverberation data are also presented.

3:35

P7. Observations on perturbation theory of scattering by surface waves. Robert H. Mellen (Kildare Corporation, 95 Trumbull Street, New London, CT 06320)

First-order perturbation theory of scattering by surface waves is usually expressed in terms of the wavenumber spectrum of the surface roughness. Bistatic reverberation is calculated from the Bragg condition and coherence loss by integration over the wavenumber range corresponding to real scattering angles. An alternative angular-spectrum approach is presented that provides additional insight into both the elementary acoustics and the nature of the approximations involved. Scattering is shown to be equivalent to dipole reradiation by a random ensemble of coherent arrays of area πL^2 , where L is correlation length. Phase delay of the excitation then produces a scatter beam pattern having horizontal broadside and vertically steered endfire components. Some of the possible hazards involved in comparing predictions with experiment are discussed.

3:50

P8. Observations of acoustic backscatter from oceanic wastewater outfalls. John R. Proni and W. Paul Dammann (National Oceanic and Atmospheric Administration, AOML-OAD, Miami, FL 33149)

Narrow beam echo sounders were used to characterize the spatial distribution of wastewater plume material from six ocean outfalls in the South Florida area. Volume scattering strengths were computed and plotted as a function of depth and horizontal distance for 200-kHz echoes. Reduction in peak scattering strength with increased range from the outfall locations correlated well with reduction in concentration of Rhodamine-WT dye introduced into the undiluted wastewater at a concentration of 1 ppm. Sound power reflection coefficients ranging from 10^{-5} to

10^{-9} were observed for these wastewater plumes. The data presented demonstrate the degree to which the sound power reflection coefficient for a distribution of scatterers is dependent upon the concentration of those scatterers, and show the utility of the acoustical method in water-mass characterization.

4:05

P9. Sound scattering by deformed cylinders of finite length. T. K. Stanton (Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

An approximate solution is derived for the scattering of sound by circular cylinders of finite length with a deformed axis and composition profile and (cross-sectional) radius that vary arbitrarily along the axis. This solution is a generalization of previous work [T. K. Stanton, J. Acoust. Soc. Am. **83**, 55–63 (1988) and T. K. Stanton, J. Acoust. Soc. Am. **83**, 64–67 (1988)] where the scattering by straight finite cylinders of uniform fluid and elastic material, respectively, were described. The new formulation is used to calculate the scattering of sound at all frequencies from a prolate spheroid with a high aspect ratio (ratio of major axis to minor axis) and a uniformly bent finite cylinder of constant cross-sectional radius. There is excellent agreement between the calculations involving the prolate spheroid and the exact spheroidal wave function solution. Furthermore, numerical integration of the deformed cylinder formula required far less computer time than calculating the exact solution. The calculations involving the bent cylinder are compared with backscatter data from preserved euphausiids (shrimplike marine organisms) and suggest that the radius of curvature of the animals plays a major role in the acoustic scatter characteristics of the marine organisms. For example, at 200 kHz, the backscattering cross section of a 23-mm-long euphausiid will decrease by 6 dB if the animal bends by as little as 1.4 mm at the ends. [Work supported by the ONR.]

4:20

P10. A comparison between the boundary element method and the superposition method for the analysis of the scattered acoustic fields from rigid bodies and elastic shells. Russel D. Miller (NKF Engineering, Inc., 12200 Sunrise Valley Drive, Reston, VA 22091), Hansen Huang (Code R14, Naval Surface Warfare Center, White Oak Silver Spring, MD 20903-5000), E. Thomas Moyer, Jr. (CMEE Department, George Washington University, Washington, DC 20064), and Herbert Überall (Department of Physics, Catholic University of America, Washington, DC 20064)

The steady-state analysis of submerged rigid and elastic bodies using two approaches is presented. In the first approach, a combined finite element/boundary element approach is used. The finite element program NASTRAN (NASA structural analysis) is used to formulate the structural matrices. The SIERRAS (surface integral equation radiated noise and analysis) program is then used to solve the coupled fluid-structure interaction problem. A superparametric boundary element with nine nodes is used. In the second approach, the superposition method is employed for modeling the fluid. The superposition method employs a number of point sources moved inside the body to represent the fluid response at the surface. This allows the fluid matrices to be formed without surface integration. Formulations for the superposition method are given for both the radiation and scattering problems. The methodologies are demonstrated for the scattering of an infinite set of plane waves from a submerged rigid sphere, prolate spheroid, and elastic spherical shell.

Session Q. Speech Communication IV: Synthesis and Coding

John G. Parker, Chairman

RADC/IRAA, Griffiss Air Force Base, New York 13441

Contributed Papers

2:45

Q1. Diagnostic tests of segmental duration models. Jan P. H. van Santen and Joseph P. Olive (AT&T Bell Laboratories, Murray Hill, NJ 07974)

Duration models map combinations of segmental identities and multifactorial prosodic contexts onto the temporal domain. Two new diagnostic tests that focus on interactions between contextual factors were applied to vowel durations measured in contexts varying in several prosodic factors and speaking rate. The first test concerns a distinction that cuts across a wide array of models (e.g., additive, multiplicative): Which groups of factors are functionally combined? When factors are functionally combined, constellations of values that yield identical durations for one segment should also yield identical durations for other segments. The data showed that speaking rate cannot be combined with other prosodic factors, contradicting any generalized version of the incompressibility model [D. H. Klatt, *J. Acoust. Soc. Am.* **59**, 1208-1221 (1976)] that would count speaking rate among its prosodic factors. The second test concerns models that express duration as the sum of additive and multiplicative terms, and can diagnose which terms are needed. Results included that speaking rate is combined with segmental identity into a single multiplicative term. These results show the advantages of diagnostic tests over the standard approach of model fitting.

2:57

Q2. A study of two standard speech intelligibility measures. Steven L. Greenspan, Raymond W. Bennett, and Ann K. Syrdal (AT&T Bell Laboratories, Naperville, IL 60566)

In the diagnostic rhyme test (DRT) participants identify each test word by choosing one of two response alternatives that differ only in their initial consonants and only by a single binary distinctive feature. Although the DRT is an accepted industry and military standard for measuring initial consonant intelligibility, other intelligibility measures may be more appropriate when stimuli produce near maximum DRT scores. Moreover, the DRT incorporates assumptions that may be valid for natural speech, but are untested for high-quality, low-bit rate coded speech. In particular, the DRT implicitly assumes that segmental intelligibility can be adequately measured by examining only single feature confusions. To examine this assumption, subjects were asked to identify the initial consonants of consonant-vowel syllables. Multifeature confusions were far more common with coded speech than with natural speech. Moreover, the consonant identification procedure reliably discriminated between speech coding devices that had near maximum, statistically indiscriminable DRT scores. These results suggest that open-response identification procedures may be more suitable than the DRT for evaluating high-quality, low-bit rate coders.

3:09

Q3. Voice conversion and its applications. Pan Jianping (Department of Information Technology in Education, East China Normal University, North Zhongshan Road, Shanghai 200062, People's Republic of China)

This approach depends on assumptions and previously proposed conclusions [D. G. Childers *et al.*, *Proc. ICASSP* **85**, 748-751]. Some corresponding voiced phonemes are extracted from the source speech and target speech to compare spectrum differences for vocal tracts of these two kinds of speech. Vocal tract lengths mainly determine spectrum distributions; they do not change greatly and are assumed to have the same varying tendencies from segment to segment while terminal radiation, etc. slightly influence spectrum distribution. These imply that there is a non-time-varying nonlinear frequency-shifting function obtained by comparing corresponding spectrum parts (especially formants) of corresponding phonemes for source and target voices. Changes of vocal tract areas based on physiological acoustics and terminal radiations for two voices are assumed to have the same tendencies for influencing the corresponding spectra from phoneme to phoneme. A non-time-varying filtering function therefore needs to be fixed by amplitude comparison between corresponding-phoneme spectra for the frequency-shifted source and target voices. Since the glottal pulse shapes remain nearly the same, and any two kinds of pitch change with the same tendencies from segment to segment, the pitch conversion factor is determined from comparing the corresponding phonemes for two voices and an excitation model whose shape is closest to that of the target voice. During voice conversion, the spectrum of the source speech is processed with the above two functions and the excitation model is impulsed by the pitch-converted target pitch from segment to segment. Areas of applications are: (a) restoring helium speech; (b) mimicking voice; and (c) preprocessing speech for recognition.

3:21

Q4. Improved duration rules for text-to-speech synthesis. Ann K. Syrdal (AT&T Bell Laboratories, IHP 1B-206, 200 Park Plaza, Naperville, IL 60566)

The durations of phonemic segments in an utterance synthesized by rule are important because they affect both intelligibility and prosodic quality. There are many factors that influence segmental duration, including the identity of the segment, and extrinsic factors such as surrounding segments, lexical stress, syntax, and speaking rate. Several approaches to modeling the combined effects of the contributory factors have been proposed. Until recently, the duration rules of the experimental AT&T dyad synthesis-by-rule system have been primarily additive. Starting with a minimum duration for each segment, duration was augmented incrementally by each contributory extrinsic factor. However, in natural speech, the effect of an extrinsic factor tends to vary depending upon other contributory factors. Analyses of natural speech [H. S. Gopal and A. K. Syrdal, *J. Acoust. Soc. Am. Suppl.* **1** **82**, S16 (1987)] indicated that Klatt's incompressibility model was more accurate than strictly additive or multiplicative approaches. Consequently, a new duration rule program was developed for the AT&T text-to-speech system, using an incompressibility approach. Comparative evaluations indicate that segment durations determined by the new duration rule system are more accurate and are preferred significantly more often by listeners than durations determined by the additive rules.

Q5. A speech synthesizer for rule-based synthesis. Yukio Mitome (C&C Information Technology Research Laboratories, NEC Corporation 4-1-1 Miyazaki, Miyamae-ku, Kawasaki, 213 Japan) and Noriko Umeda (Department of Linguistics, New York University, 719 Broadway, Room 505, New York, NY 10003)

A new speech synthesizer is proposed that is suitable for speech synthesis by rule. This synthesizer is based on a source-filter model and consists of two sources—a glottal source and a noise source—and three cascade filters. Each source can be connected with any of the filters dynamically. Each filter is used as a speech component, which is produced as a source signal filtered by a slow varying system and has a limited duration. The speech signal consists of some components overlapping each other in time-frequency domain. The experimental system was developed on a 32-bit personal computer. The software system has two subsystems, a speech synthesizer with a format pattern interpreter and a speech analysis system with a speech database. A formant pattern interpreter reads a formant pattern definition file, and a controls speech synthesizer routine. Speech analysis is based on the maximum entropy method (MEM), and a database contains 25 min of recorded male voice samples. Experimental results showed that this synthesizer can easily realize the spectral discontinuity between phonemes.

3:45

Q6. Improved excitation prediction and quantization in optimal amplitude multipulse coders. Daniel Lin (International Mobile Machines Corporation, 2130 Arch Street, Philadelphia, PA 19103)

Improvements to the excitation prediction and quantization procedure in optimal amplitude multipulse coders are described. The new procedure successively reoptimizes the pulse amplitudes and predictor gain of the long-delay correlation filter as each new pulse is found. Furthermore, the amplitudes and gain are requantized at each step so that the new pulse amplitudes and locations can correct for the quantization errors in the existing excitation. To perform the joint amplitude-predictor reoptimization, nonrecursive pitch prediction structure is used in our analysis. The predictor is implemented as an adaptive vector quantizer whose codebook is populated with past excitation sequences. This structure allows for a computationally efficient closed-loop analysis procedure. The results showed that at 9.6 kbps, the new technique achieved an average segmental SNR of over 18.5 dB. Informal subjective tests indicated that the reconstructed speech is toll quality.

Q7. TFT: An algorithm for the spectral compression of natural speech signals. Richard R. Hurtig (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

An earlier report [R. R. Hurtig, J. Acoust. Soc. Am. Suppl. 1 81, S78 (1986)] using synthesized syllables demonstrated that naive subjects can discriminate and identify spectrally compressed vowel segments under auditory and vibrotactile conditions. These findings are consistent with the view that the identification of the spectral shape of the speech segment may be independent of its frequency range. A computational algorithm was developed to achieve spectral compression of natural speech. The algorithm includes calculation of an n -point FFT, padding the result with the spectrum of a Hamming window, calculation of a $2n$ -point IFFT, and outputting the first half of the resultant time domain signal. The size of the pad determines the amount of compression achieved while the placement of the pad determines the direction of the frequency shift. Naive subjects had no difficulty recognizing simple sentences in a closed set for speech signals compressed to 2500 or 1250 Hz bands. After a few hours of listening, open set recognition is achieved. The implementation of the algorithm for sensory aids will be demonstrated.

4:09

Q8. Statistical tree-based modeling of phonetic segment durations. Michael D. Riley (Department of Linguistics, AT&T Bell Laboratories, Murray Hill, NJ 07974)

Segmental durations are affected by many factors: phonetic context, speaking rate, stress, word and phrasal position, etc. Regression trees [L. Breiman *et al.*, *Classification and Regression Trees* (Wadsworth, Monterey, CA, 1984)] are well suited to capturing these effects, since they (1) statistically select the most significant features, (2) permit both categorical and continuous factors to be considered, (3) provide "honest" estimates of their performance, and (4) allow human interpretation and exploration of their result. In particular, transcribed databases of 400 utterances from a single speaker and 4000 utterances from 400 speakers of American English were used to build optimal decision trees that predict segment durations based on such factors. Over 70% of the durational variance for the single speaker and over 60% for the multiple speakers were accounted for by this method when using information only at the word level and below. These terms were used to derive durations for a text-to-speech synthesizer and were found to give more faithful results than the existing heuristically derived duration rules. Since tree building and evaluation is rapid once the data are collected and the candidate features specified, the technique can be readily applied to other feature sets and to other languages.

Session R. Noise IV: Environmental Noise and Impact on Hearing

David Lubman, Chairman

*Hughes Aircraft, Building 618, MS H425, Post Office Box 3310, Fullerton, California 92631***Contributed Papers****8:00**

R1. Canadian "National Guidelines for Environmental Noise Control—Procedures and Concepts for the Drafting of Environmental Noise Regulations/By-laws in Canada." Deirdre A. Morison (Bureau of Radiation and Medical Devices, National Health and Welfare, Canada, Ottawa, Ontario K1A 0L2, Canada)

These National Guidelines have been prepared for legislators at all levels of government, provincial planners, municipalities, consultants, industries, and designers. The intent is to provide a common basis across Canada for the assessment, measurement, and legislative control of environmental noise while, at the same time, providing options to allow flexibility of choice to fit specific needs. The National Guidelines may be adopted or modified, in entirety and in part, into provincial or municipal legislation or into codes of practice. The National Guidelines are divided into two major parts. Part I, Concepts and Procedures, details the various options available in developing a noise control program and includes a section on Land Use Planning and Model Noise Control Legislation presenting sound level objectives and more general bylaws. The second part of the document contains technical reference material, including a section on instrument specification, measurement, and prediction, and another covering noise reduction techniques. Terms and interpretations, references, and technical support documents are included also. The texts of the technical support documents are briefly summarized in the document and are reproduced on microfiche at the end of the National Guidelines. The National Guidelines were prepared by the Working Group on Environmental Noise on the Federal/Provincial Advisory Committee on Environmental and Occupational Health, which intends to provide periodic revisions.

8:15

R2. Public reaction to low levels of aircraft noise. John E. Wesler (Wyle Laboratories, 2001 Jefferson Davis Highway, Arlington, VA 22202)

Several recent instances have raised the issue of public annoyance from the noise of airplanes flying at relatively high altitudes or at relatively large distances from the nearest airport. Public complaints have arisen about airplane flights over northern New Jersey as the result of changes in flight patterns associated with the major New York airports, even though in many instances those airplanes are flying at 15 000 ft or higher. Concerns have arisen regarding the noise levels on the ground from the new, swept-blade, advanced turboprop airplanes when they are flying at cruise altitudes of 30 000 ft and higher. Complaints about aircraft noise over national parks have resulted in a Congressional requirement to measure those noises and determine their severity. These noise levels do not meet the usual criteria for annoyance or interference with individual activity, whether in terms of average level or single events. A better understanding of the intrusive effects of low levels of community noise is needed, especially where present in areas of relatively low ambient noise levels.

8:30

R3. The noise environment under low-altitude, high-speed military aircraft training routes. Kenneth J. Plotkin (Wyle Laboratories, 2001

Jefferson Davis Highway, Suite 701, Arlington, VA 22202) and Alton Chavis (HQ TAC/DEEV, Langley Air Force Base, VA 23665)

Low-altitude, high-speed training operations are routinely conducted along specially designated Military Training Routes (MTRs). Design of new routes and/or realignment of existing routes requires an environmental assessment to determine the community noise impact. Nominally, aircraft on these routes navigate from point to point along defined segments. Key elements required for noise prediction are the frequency of flights, the statistical variation of position relative to the nominal centerline, and the operational noise emission levels of the aircraft. Noise measurements were conducted on three routes: one operated by the Strategic Air Command and two by the Tactical Air Command. On each route, 20 automatic noise monitors were deployed on a 2- to 4-mi array across the route centerline. Major findings were: noise emission levels are fully consistent with predictions from USAF's NOISEFILE database; aircraft tend to fly within the central part of a route; aircraft follow nominal tracks corresponding to defined route centerline or corresponding to prominent visual references; the lateral distribution about each track is Gaussian; and multiple tracks can exist. Noise events were infrequent (typically less than three or four per day), and the highest L_{dn} measured was less than 65 dB. [This work was sponsored by USAF AAMRL/BBE.]

8:45

R4. ROUTEMAP model for predicting aircraft noise exposure along military training routes. Michael J. Lucas and Kenneth J. Plotkin (Wyle Laboratories, 2001 Jefferson Davis Highway, Suite 701, Arlington, VA 22202)

A model and PC-based computer program has been prepared to calculate noise levels along low-altitude, high-speed military training routes. The program is designed for use by environmental planning personnel who are familiar with MTR operations and with noise, but are not necessarily expert. The program provides options for selecting general types of operations (visual or instrument navigation), aircraft types and speeds, altitudes, and nominal track centerlines. Up to 20 track/altitude/aircraft types may be defined within a 20-mi-wide corridor. Aircraft on each track have a Gaussian lateral distribution about the centerline. The program contains nominal standard deviations based on the type of operations, or the user may specify a site-specific value. Aircraft noise emission levels are derived from the USAF NOISEFILE database. The program calculates L_{eq} , L_{dn} , and L_{dnmr} , where L_{dnmr} is L_{dn} with an adjustment to account for the onset rate of MTR aircraft noise. Program output is available in tabular form or in graphs suitable for inclusion in reports. [This work was sponsored by USAF AAMRL/BBE.]

9:00

R5. Sonic boom spectra of Atlantis landing 6 December 1988. Robert W. Young and Frank T. Awbrey (Sea World Research Institute, Hubbs Marine Research Center, 1700 South Shores Drive, San Diego, CA 92109)

After circling the world 69 times, orbiter Atlantis came to a stop on Runway 17 of Edwards Air Force Base in California, at 1537U on 6

December 1988. Some 10 mi to the west, and 4 min previous to landing, its 394-ms sonic boom swept over our measurement site at latitude 34.886°N, longitude 118.036°W, elevation about 2.5 kft above sea level. When the decelerating Atlantis heading southeast passed nearest to the microphones at lateral slant range of 100 kft, it was gliding at Mach 1 about 48 kft above ground. The peak flat sound pressure level of the sonic boom was 129 dB; the peak C-weighted sound pressure level, 125 dB; the peak A-weighted sound pressure level, 110 dB. The flat sound exposure level was 118 dB; the C-weighted and A-weighted sound exposure levels of the initial transients were, respectively, 102 and 84 dB. If sound exposure level is wanted for a 400-ms sonic boom with primary emphasis in the range 0.8–3 Hz, flat sound exposure level is appropriate; with primary emphasis in the range 10–50 Hz, C-weighted sound exposure level is appropriate; with primary emphasis in the range 80–1000 Hz, A-weighted sound exposure level is appropriate.

9:15

R6. Revision of a dosage-effect relationship for the prevalence of noise-related annoyance. Sanford Fidell (BBN Systems and Technology, Corporation, P. O. Box 633, Canoga Park, CA 91304)

More than a decade has passed since Schultz (1978) derived a relationship between noise exposure and the prevalence of annoyance from the findings of various social surveys of community response to general transportation noise sources. Numerous measurements of the prevalence of noise-related annoyance have been published in subsequent years. A least-squares quadratic fitting function to a set of 427 data points (developed by addition of 266 new data points to the original set of 161 data points) differs little from the third-order polynomial fit to the original data set. [Work supported by U.S. Air Force Noise and Sonic Boom Impact Technology Program under Contract F33615-86-C-0530.]

9:30–9:45

Break

9:45

R7. Effects of changing the A-weighting design goal. George S. K. Wong (Division of Physics, National Research Council, Ottawa, Ontario K1A 0R6, Canada)

The proposal by the International Organization for Standardization (ISO), document ISO/TC43/SC1 N644, dated October 1988, to modify the A-weighted design goal by imposing additional attenuation to implement well-defined frequency cutoffs at 20 Hz and at 16 kHz to ensure consistent A-weighted measurements is examined. The merit of the above proposal is in doubt since the high-frequency cutoff at 16 kHz eliminates both audible and high-frequency components and can result in underestimation during A-weighted noise assessment, particularly when the noise is impulsive. The consequences of the above changes are serious: Future databases of A-weighted measurements will be incompatible with those from the past, and acoustical communities in every country will suffer financial loss due to the need to replace most of their measuring instruments to comply with the proposed A-weighting. A better approach to ensure consistent A-weighted measurement is to impose tighter tolerances in the high-frequency region of the A-weighting, such as those specified in ANSI S1.4A-1985 amendment to ANSI S1.4-1983.

thresholds at 2–4 kHz in the left ear than in the right ear. Measured L_{eq} correlated with HTLs at 3–4 kHz. Playing in a symphony orchestra appears to present a mild risk of hearing damage, but musicians in every age group displayed average hearing thresholds better than the general population.

10:15

R9. Abstract withdrawn.

10:00

R8. Sound exposures and hearing thresholds of musicians in a major symphony orchestra. Julia Doswell Royster (Environmental Noise Consultants, Incorporated, P. O. Box 144, Cary, NC 27512-0144), Larry H. Royster (North Carolina State University, Raleigh, NC 27695), and Mead C. Killion (Etymotic Research, Elk Grove, IL 60007)

Seventy noise dosimetry samples were obtained for musicians during rehearsals and performances of a major symphony orchestra. Audiograms were obtained for 59 musicians. The L_{eq} during measurement periods ranged from 76–102 dBA (median = 90 dBA), corresponding to on-the-job daily equivalent values of 72–98 dBA (median = 86 dBA). Using the ISO 1999.2 model, this exposure would be expected to produce 5–8 dB of NIPTS after 30 yr for typical ears (0.5 fractile) or 8–10 dB of NIPTS for very susceptible ears (0.05 fractile). The musicians' average thresholds were better than those for age-matched reference nonindustrial noise-exposed populations without occupational noise exposure, and only slightly worse than those for highly screened populations representing aging alone. However, audiogram patterns indicated a slight notch, suggesting a contribution from NIPTS. Bilaterally averaged thresholds for musicians in different instrument sections were essentially equivalent, but violinists and violists grouped together showed significantly poorer

10:30

R10. Review of the effects of noise on performance. Alice H. Suter (Bioacoustics and Occupational Vibration Section, MS C-27, Division of Biomedical and Behavioral Science, NIOSH, 4676 Columbia Parkway, Cincinnati, OH 45226)

The effects of noise on job performance are not as easily discerned and predictable as other effects, such as those on hearing or speech communication. The extent to which noise affects performance depends on numerous nonacoustical factors, such as the subject's biological and psychological state, as well as certain external factors. Despite these and other difficulties involved in comparing research results, a recent review of the

noise and performance literature provides sufficient evidence to indicate adverse effects under certain circumstances. The probability of performance decrements increases with increased (1) noise level, (2) intermittency, (3) aperiodicity, (4) lack of controllability, (5) task complexity, (6) task duration, and (7) the addition of certain other stressors. The review was sponsored by the U.S. Army Human Engineering Laboratory, and performed under the auspices of Gallaudet University.

10:45

R11. Growth of threshold shift from intense impulses: Implications for basic loss mechanisms. G. Richard Price (US Army Human Engineering Laboratory, Aberdeen Proving Ground, MD 21005-5001)

For a given number of impulses, increases in intensity are normally associated with greater threshold shifts. The details of this growth should, in principle, reveal something of the basic mechanisms responsible for the loss. Experiments have been conducted in which the ears of 30 cats have been exposed to 50 impulses with their peak energies located at 4000 Hz and with peak pressures ranging from 135–145 dB SPL. Threshold shifts grew about 7.0 dB for every dB increase in SPL above 134 dB. Virtually the same result can be calculated from data for the chinchilla ear [R. P. Hamernik, J. H. Patterson, and R. J. Salvi, *J. Acoust. Soc. Am.* **81**, 1118–1129 (1987)], which may indicate that this function is characteristic of mammalian ears. On the other hand, exposures to other impulses in both animals have shown much lower rates of growth; however, these differences can be explained by conductive nonlinearities in the middle ear and/or a limited range of growth of threshold shift.

WEDNESDAY MORNING, 24 MAY 1989

REGENCY A, 8:00 TO 10:35 A.M.

Session S. Physical Acoustics III and Engineering Acoustics III: Recent Advances in Thermoacoustic Engines

Thomas J. Hofler, Chairman
Physics Department, Code 61HF, Naval Postgraduate School, Monterey, California 93943

Chairman's Introduction—8:00

Invited Papers

8:05

S1. A thermoacoustically driven orifice-pulse-tube cryocooler. G. W. Swift (Condensed Matter and Thermal Physics, Los Alamos National Laboratory, Los Alamos, NM 87545), R. A. Martin (Advanced Engineering Technology, Los Alamos National Laboratory, Los Alamos, NM 87545), and Ray Radebaugh (National Institute for Standards and Technology, Boulder, CO 80303)

The orifice pulse tube is a variant of Stirling-cycle refrigerator in which the cold, work-absorbing piston and crankshaft are replaced by a simple dissipative structure. To date, the oscillatory pressure needed to drive pulse tubes has been provided by complex, unreliable room-temperature pistons and crankshafts. In this presentation, plans and design calculations for driving an orifice pulse tube with a thermoacoustic engine will be discussed. In this composite device, heat flowing from a high-temperature source to a room-temperature sink will generate high-amplitude acoustic oscillations in high-pressure helium gas; these oscillations will power the orifice pulse tube, thereby pumping heat from a low-temperature source to a room-temperature sink. Thus the device is a heat-driven cryocooler with no moving parts. In this planned small laboratory device, it is expected that a few W of refrigeration at roughly 80 K using about 2 kW of heat at 1000 K will be produced. The nature of the oscillatory heat transfer and time-averaged enthalpy flow in the various components of the engine and cooler will also be discussed, comparing and contrasting the acoustic-frequency Stirling cycle in the cooler with the thermoacoustic cycle in the engine.

8:35

S2. The dependence of the experimental performance of a thermoacoustic refrigerator on geometry. T. Hofler and M. Suzalla (Department of Physics, Naval Postgraduate School, Monterey, CA 93943)

The results of recent measurements on an experimental thermoacoustic refrigerator will be presented. In simple terms, thermoacoustic heat transport is caused by the interaction of a high-amplitude standing wave with a stationary "stack" (a stack of parallel plates having uniform plate spacing). The focus of these measurements is on the effect changes in stack geometry has on refrigerator performance as determined by temperature

span and thermodynamic efficiency. The experimental geometry variations consist of changing the position of the stack in the standing wave and changing the spacing between adjacent plates in the stack. The results show that refrigerator performance is very sensitive to the location of the stack in the standing wave, as expected from the theory of N. Rott, but performance is much more sensitive to plate spacing than predicted by the theory. [Work supported by the Office of Naval Research and the Office of Naval Technology.]

9:05

S3. Optimizing impedances in the traveling wave heat engine. Peter. H. Ceperley (Departments of Physics and Electrical and Computer Engineering, George Mason University, Fairfax, VA 22030)

Thermal coupling between solid surfaces and an acoustic working fluid in a regenerator is essential to the operation of a thermoacoustic engine, while the associated viscous coupling, also between the surfaces and the working fluid, is detrimental to the engine's performance and efficiency. The Prandtl number (roughly 0.7 for most gases over a considerable range of temperatures and pressures) fixes the ratio of these two types of coupling for a thermoacoustic engine with a continuous regenerator. This ratio of couplings can be improved in several ways in traveling wave heat engines with short regenerators [P. H. Ceperley, *J. Acoust. Soc. Am.* **66**, 1508-1513 (1979) and P. H. Ceperley, *J. Acoust. Soc. Am.* **77**, 1239-1244 (1985)]. This paper will discuss the optimization of the flow resistance and length of a short regenerator and also resonant impedance enhancement using split mode excitation [P. H. Ceperley, U.S. Patent 4,686,407 (1987)] to significantly reduce the viscous losses while maintaining good thermal coupling.

9:35

S4. Vortices as a natural aeroacoustic engine. M. Kurosaka (Department of Aeronautics and Astronautics, University of Washington, Seattle, WA 98195)

In previous investigations, it was found that the so-called Ranque-Hilsch effect is induced by a spinning acoustic wave present in swirling flows; the total temperature within the vortices becomes acoustically separated. Another similar example for the Karman vortex street or an array of vortices formed a bluff body is presented here. The vortices in the wake are found to have the capacity to separate the total temperature, and the separation is significantly enhanced by acoustic resonance. Here, vortices behave as a natural "compressor-turbine" cycle, its performance improved substantially by acoustics.

Contributed Papers

10:05

S5. Thermoacoustic refrigerator for space applications. S. L. Garrett, T. Hoffer, M. Fitzpatrick, M. P. Susalla, R. Volkert, D. Harris (Physics Department, Code 61 Gx, Naval Postgraduate School, Monterey, CA 93943), R. B. Byrnes, C. B. Cameron (Space Systems Academic Group, Code 66, Naval Postgraduate School, Monterey, CA 93943), and F. M. Murray (JBL, Incorporated, Northridge, CA 91324)

Long-lived, space-based cryocoolers with low vibration levels are necessary in a variety of applications, including the cooling of infrared sensors and high T_c superconductors. A thermoacoustic refrigerator designed to function autonomously in a Space Shuttle Get Away Special (GAS) canister will be described that has no sliding seals and only 15 gm of reciprocating mass. It is capable of cooling to 100 K below room temperature with a single stage "stack" consisting of rolled plastic film and short lengths of 10-lb test monofilament fishing line as the spacers. This presentation will concentrate on the modifications to the basic refrigerator design (U.S. Patent No. 4,722,201, issued 2 Feb. 1988) that improve both its electroacoustic and thermoacoustic efficiencies. These improvements include the use of a helium-xenon gas mixture as the thermodynamic working fluid and a neodymium-iron-boron electrodynamic driver with a titanium suspension. Custom electronic circuits necessary to control the refrigerator (microprocessor, bubble memory data recorder, phase-locked loops, automatic gain controls, amplifiers, multiplexed diode thermometers, etc.) and special fabrication techniques necessary to

confine the helium for years while permitting electrical feed through will also be disclosed. [Work supported by the Office of Naval Research, Office of Naval Technology, and the Naval Postgraduate School Research Foundation.]

10:20

S6. Source level limits for submerged thermoacoustic sound sources. Thomas B. Gabrielson (Naval Air Development Center, Code 5044, Warminster, PA 18974)

One application for thermoacoustic engines is the generation of sound underwater. In theory, the engine resonator could either be flooded with water and radiate directly into the surrounding water, or the resonator could be gas-filled. The radiation performance of a liquid-filled resonator (taken to be an open-ended tube) radiating into the same liquid is well known; however, the open end of a submerged, gas-filled tube is almost a velocity node, so a smaller fraction of the stored energy "leaks" out as radiation. For the same pressure amplitude in both the liquid-filled and gas-filled tubes, the particle speed is much higher in the gas-filled tube, and this more than compensates for the smaller transmission coefficient into the surrounding liquid. In either case, limits to the acoustic power output can be calculated from the resonator radiation efficiency and from the peculiar requirements of the thermoacoustic engine.

Session T. Underwater Acoustics III and Structural Acoustics and Vibration III: Commonality Between the Fields of Underwater Acoustics and Structural Acoustics

John J. McCoy, Chairman

School of Engineering and Architecture, Catholic University of America, Washington, DC 20064

Chairman's Introduction—8:00

Invited Papers

8:05

T1. Radiation and scattering from laminated spherical shells. Henrik Schmidt (Massachusetts Institute of Technology, Cambridge, MA 02139)

The solution in terms of spherical harmonics to acoustic and elastic wave propagation problems in spherically stratified media is well established. However, except for very low ka values, a direct numerical implementation is unstable. Propagator matrix approaches have been applied for moderate ka values, whereas environment transformations have been developed to represent the spherical stratification by a plane stratification at higher ka values, in particular in relation to long-range seismic propagation in the solid earth. Here, it is demonstrated that, by introducing a proper normalization of the spherical Bessel functions, unconditional stability can be obtained by using the Global Matrix approach [H. Schmidt and F. B. Jensen, *J. Acoust. Soc. Am.* **77**, 813–825 (1985)]. This SAFARI code has therefore been modified to treat propagation in spherically stratified elastic media. The algorithm is stable at least up to $ka = 10^3$, allowing modeling of stratifications alternating between high- and low-speed layers, traditionally problematic for propagator approaches. Examples will be given for high-frequency scattering from coated shells as well as simulations of array signal processing performance in relation to structural acoustics experiments. [Work supported by ONR.]

8:30

T2. Three-dimensional Green's function for fluid-loaded thin elastic cylindrical shell: Formulation and solution. L. B. Felsen, J. M. He, and I. T. Lu (Department of Electrical Engineering/Computer Science, Weber Research Institute, Polytechnic University, Farmingdale, NY 11734)

This paper treats sound radiation from a time-harmonic point pressure source located either inside or outside a thin, homogeneous, infinitely long circular cylindrical elastic shell, which is immersed in different interior and exterior fluid media. This Green's function problem is attacked by a combination of the method of separation of variables and the method of images applied to an infinitely extended azimuthal (ϕ) domain. The reduced one-dimensional problems in the cylindrical (r, ϕ, z) coordinates are solved by general spectral techniques in terms of one-dimensional characteristic Green's functions g_r, g_ϕ, g_z , which depend on one or both of the two complex spectral separation parameters (spatial wavenumbers) λ_1 and λ_2 . While the one-dimensional problems in the ϕ and z domains are straightforward, the presence of the shell in the radial domain introduces substantial complexity. The solution is obtained by defining the discontinuities in the pressure and normal displacement across the shell via recourse to the dynamical equations of motion inside the shell. The synthesis problem is made unique through a complete analysis of the spectral singularities $g_{r,\phi,z}$ in their respective complex planes, which permits selection of appropriate integration contours. A host of alternative representations, whose choice (concerning utility) is motivated by the parameter range of interest, can be derived from the fundamental spectral form, and asymptotic reductions lead to a variety of wave processes that have a cogent ray acoustic interpretation. [Work supported by ONR.]

8:55

T3. Coherence theory in volume scattering and structural acoustics. Mark J. Beran and John J. McCoy (Department of Civil Engineering, Catholic University of America, Washington, DC 20064)

In this paper, the use of the two-point coherence function, defined for arbitrarily positioned points, to study wide-angle volume scattering problems in the ocean and wave propagation in cylindrical shells subject to random forcing is considered. Although the basic equations of the two phenomena are different, it will be shown that the basic approach developed 30 years ago in optics is a useful way to study these and similar problems. First, the theory of partial coherence developed in optics for free-space propagation from random sources is reviewed. Then, the coherence equations for the cylindrical shell propagation case are formulated, and it is shown how the equations may be solved when the two-point statistics of the forcing function are known.

9:20

T4. Frequency wavenumber analysis of seismoacoustic waves in an ice layer. G. Giellis and T. C. Yang (Naval Research Laboratory, Code 5123, Washington, DC 20375)

Seismoacoustic waves traveling in an ice layer over a deep Arctic Ocean are studied using frequency wavenumber analysis. Ice-ridge-generated noise can travel in the ice via ice-trapped waves and water-borne waves coupled to the ice. {Previous hydrophone data [B. Buck and J. H. Wilson, *J. Acoust. Soc. Am.* **80**, 256-264 (1986)] indicated that the noise originated from the bottom of a ridge.} The data are simulated by a point source in the water (for the water-borne waves) and a point source in the ice (for the ice-trapped noise) using the SAFARI numerical code with a planar receiver array of vertical axis geophones. Frequency wavenumber analysis is applied to the simulated data to determine the wavenumber of the various waves traveling in the ice. The methodology and preliminary results of this analysis will be reported.

9:35

T5. Low-frequency diffraction from a free surface coupled to a semi-infinite elastic surface as modeled by sea ice properties. Peter H. Dahl (MIT/WHOI Joint Program in Oceanography/Oceanographic Engineering, Department of Ocean Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139) and George V. Frisk (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

This paper discusses the solution of a low-frequency plane wave incident upon a semi-infinite elastic plate, such as an Arctic ice lead or free edge, using the Wiener-Hopf method. By low-frequency it is meant that the elastic properties of the plate are adequately described by the thin plate equation. For example, in a floating ice sheet, this translates into frequency-ice thickness products that are $\lesssim 150$. A key issue here is the fluid

loading pertaining to sea ice and low-frequency acoustics, which cannot be characterized by simplifying heavy or light fluid loading limits. An approximation to the exact kernel of the Wiener-Hopf functional equation is used here, which is valid in this midrange fluid loading regime. The farfield diffracted pressure is found, which includes a fluid-loaded, subsonic (relative to the water) flexural wave in the ice plate. Comparisons are also made with the locally reacting approximation to the input impedance of an ice plate. The combined effects of the ice lead diffraction process represent loss mechanisms that contribute to the transmission loss in long-range Arctic acoustic propagation.

9:50

T6. Low-frequency SAW sensors for the detection of shear stresses in the turbulent boundary layer. Yongrae Roh, Vasundara V. Varadan, and Vijay K. Varadan (Department of Engineering Science and Mechanics, Research Center for the Engineering of Electronic and Acoustic Materials, The Pennsylvania State University, University Park, PA 16802)

To simulate a low-frequency SAW device, surface wave propagation at the boundary between water and a thin film of PZT-5H on steel was investigated extensively through numerical analysis. Two- and three-dimensional dispersion curves, attenuation mechanism, and displacement variation were obtained for each of the propagation modes of all types of the surface waves (Rayleigh, Scholte, and Love) in the medium. The energy distributions of the Rayleigh and Scholte waves have also been obtained. With these results, the optimum geometry (crystal cut, propagation direction, and nondimensional wavenumber) for maximum launching efficiency was determined. The acoustic fluid was replaced by a turbulent flow, and the variation of the propagation velocity of the SAW due to turbulence was investigated. These results show a new method to distinguish the effect of shear stress fluctuations from that of normal pressure in a turbulent flow.

WEDNESDAY MORNING, 24 MAY 1989 GOLDSTEIN AUDITORIUM, 8:15 A.M. TO 12:00 NOON

Session U. Speech Communication V: Perception, Production, Analysis, Synthesis, and Coding (Poster Session)

Richard Pastore, Chairman

Department of Psychology, SUNY at Binghamton, Binghamton, New York 13901

Contributed Papers

All posters will be on display from 8:15 a.m. to 12:00 noon. (Viewing hours will also be extended into the evening and through noon on Thursday.) To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:15 to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 to 11:45 a.m. Contributors are encouraged to leave their posters in place until 12:00 noon on Thursday. A cash bar will be set up in the evening to facilitate informal discussion. The Goldstein Auditorium will be closed at 10:00 p.m. and will reopen on Thursday morning.

U1. Abstract withdrawn.

U2. Formant contour estimation in noise using a spectral estimation "confidence measure." T. V. Sreenivas and R. J. Niederjohn (Department of Electrical, Computer, and Biomedical Engineering, Marquette University, Milwaukee, WI 53233)

Estimation of formant contours from continuous speech is a difficult problem that has required either elaborate heuristic rules or complex hidden Markov models. Under high noise levels, these methods deteriorate significantly because of the degraded performance of the spectral estimator on which they are based. This paper proposes a method of formant contour estimation using a measure of confidence for the spectral peak estimation. At high noise levels, most spectral estimation techniques result in large variance and increased spurious/missing peaks; however, it is

found that a broad distribution of spectral energy can be determined more reliably using filter-bank analysis. This spectral energy distribution is utilized to differentiate regions of high and low signal-to-noise ratio in the spectrum (for white noise corruption), which in turn is used to determine a "confidence measure" for the spectral peak estimation. The addition of this confidence information to the noisy spectral peak data of a speech utterance results in clear segments of the formant contour standing out from the remaining noisy peak data. Using these segments as anchor regions, an algorithm is developed to determine the complete contours, within the constraints of the known speech properties. [Work supported by NSF.]

U3. Duration effects on vowel perception by hearing-impaired listeners. J. Besing, M. J. Collins, and J. Cullen (Division of Communication Disorders, Louisiana State University, Baton Rouge, LA 70803)

The effect of temporal differences on the identification of vowel tokens by hearing-impaired listeners was studied using six hearing-impaired listeners. Stimuli were synthetically generated and varied in duration of steady state, durations of the initial and final transitions, and F_1 , F_2 , and F_3 location. Listeners were required to identify the given vowel token from a set of ten possible alternatives. In some cases, the hearing-impaired listeners' responses were similar to a group of normal-hearing listeners and, in some cases, they were very different and appeared somewhat idiosyncratic. The hearing-impaired subjects were different from each other; further, the labeling performance was different within the same subjects across different vowels. In addition, subjects with similar audiograms performed differently from one another on some vowels and similarly on others. These results and the particular effects of the alterations in the durations of the various components will be discussed. [Work supported by NIMH.]

U4. Linear transformations for vowel normalization. Stephen A. Zahorian and Amir J. Jagharghi (Department of Electrical and Computer Engineering, Old Dominion University, Norfolk, VA 23529)

The results of an evaluation of multivariable linear regression techniques for speaker normalization of vowel data for 11 vowel classes will be presented. The database for the study consisted of the central vowel portions of 2922 CVC syllables obtained from ten males, ten females, and ten children. Each stimulus was represented both by three formants and in terms of overall spectral shape, via the discrete cosine transform coefficients (DCTCs) of the magnitude spectra. In all classification experiments, half the database was used to train the classifier and the other half was used for evaluation. For the case of formants, the classification accuracy on the evaluation data was 63.3% if different speakers were used for training and testing (and thus no speaker normalization), 63.9% if the same speakers were used in the training and test sets but without explicit normalization, and 75.2% with speaker-normalized data. The rates for the corresponding conditions, but with DCTCs as parameters, were 58.0%, 66.1%, and 77.9%. These results indicate a reduction in classification error rate due to speaker normalization of 32.4% for the case of formants and 47.3% for the case of DCTCs. The speaker-normalization techniques are an extension of the techniques reported in the literature [S. F. Disner, *J. Acoust. Soc. Am.* 67, 253-261 (1980)]. These results also extend our results reported at the Fall 1987 ASA Meeting [S. A. Zahorian and A. J. Jagharghi, *J. Acoust. Soc. Am. Suppl.* 1 82, S37 (1987)] to a larger, more varied database and indicate that automatic classification of speaker-normalized vowels is comparable to either spectral shape parameters or formants as initial features. [Work supported by NSF.]

U5. F_0 normalization and talker variability. Keith Johnson (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Fundamental frequency (F_0) normalization is an effect observed in vowel perception in which the F_0 of the vowel plays a role in perceived vowel quality. It was hypothesized that this perceptual effect involves a process in which hearers adjust a perceptual vowel space when they detect a change of speaker. A seven-step vowel continuum from "hood" to "hud" was synthesized at two levels of F_0 (120 and 240 Hz). These two continua were presented to subjects blocked by F_0 level and with items of different F_0 randomly intermixed with each other. In the mixed F_0 condition, there was a large shift in the identification functions of the vowel continuum as a result of changing F_0 , while there was no effect of F_0 in the blocked F_0 condition. Analysis of the reaction time data showed that subjects identified the items more slowly in the mixed condition than in the blocked condition and that this increase occurred for items that were preceded by an item with different F_0 and not when successive items had the same F_0 . These data suggest that hearers adjust a perceptual vowel space when they detect a change of speaker. [Work supported by NIH Training Center Grant No. 50-315-26 to Indiana University.]

U6. Target zones for synthetic vowels. James D. Miller and John W. Hawks (Central Institute for the Deaf, 818 South Euclid, St. Louis, MO 63110)

Using algorithms developed by the second author (JWH), he synthesized 7703 vowel tokens such that the values of F_0 , F_1 , F_2 , and F_3 uniformly sampled locations in a plane in the middle of the vowel slab of the auditory-perceptual space [J. D. Miller, "Auditory-perceptual interpretation of the vowel," *J. Acoust. Soc. Am.* (in press)]. Each token was 400 ms and had a rise-fall pitch contour. The tokens were arranged in random order and presented in groups of 100 to one listener (JDM), who identified each token as [IY, IH, EH, EY, AE, AA, AH, AO, OW, UH, or UW] and rated the clarity of each token on a scale from 1 (poor) to 5 (excellent). The sizes and shapes of the target zones derived from these experiments will be compared to those based on measurements of natural speech. The data are, in general, consistent with the concept of vowel target zones as being large with irregular and abutting boundaries. [Work supported by NINCDS.]

U7. Detection thresholds for isolated vowels. Diane Kewley-Port (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

A series of experiments on the detectability of vowels in isolation has been completed. Stimuli consisted of three sets of ten vowels, one synthetic, one from a male talker, and one from a female talker. Vowel durations ranged from 20-160 ms. Thresholds for detecting the vowels in isolation were obtained from well-trained, normal-hearing listeners using an adaptive-tracking paradigm. Detection thresholds for vowels calibrated for equal sound pressure at the earphones differed by as much as 20 dB across vowels. However, the same patterns of thresholds obtained for the vowel set were obtained for all vowel durations. An orderly decrease in vowel thresholds was obtained for increased duration, as predicted from temporal integration for pure tones. Several different analyses have been conducted in an attempt to explain the differential detectability for the various vowels within each set. The most detailed of these analysis involved a model developed by Moore and Glasberg (1987) for calculating excitation patterns and the corresponding loudness level in phons. While that model provided improved explanatory power over other models tried, further refinements of perceptual models of loudness will be needed to explain these data. [Research supported by NIH and AFOSR.]

U8. Perceived modulation magnitudes of second-formant, center-frequency variations. L. S. Sullivan, R. J. Porter (Kresge Hearing Research Laboratory, LSU Medical Center, New Orleans, LA 70112 and Department of Psychology, University of New Orleans, New Orleans, LA 70112), J. K. Cullen, and M. J. Collins (Kresge Hearing Research

Laboratory, LSU Medical Center, New Orleans, LA 70112 and Department of Speech and Hearing, LSU Medical Center, New Orleans, LA 70112)

Speech signals convey information via rapid changes in formant frequencies. Listeners can reliably judge the relative "modulation magnitude" of frequency modulations [e.g., H. Fastl, *Hear. Res.*, 59-69 (1982)]. Listeners were asked if they could also judge modulation magnitude for center-frequency-modulated formants by presenting pairs of 500-ms stimuli with centered, "notchlike" modulations drawn from synthetic continua with varying notch duration (30-300 ms) or depth (50-500 Hz). The first, reference, stimulus (either a low- or high-modulation extreme of a continuum) was assigned a reference magnitude (1 for low-modulation-reference trial blocks; 10 for high-). Listeners assigned a relative "modulation strength" (1 to 10) to the second stimulus. Judged magnitude appears as an uneven perceptual surface in the magnitude \times depth \times duration space: Highest modulation strengths are heard for deep-long stimuli; deep-short stimuli have slightly lower judged magnitudes; shallow-long modulations are much weaker; shallow-short modulations are the weakest. The possible relation between perceived modulation magnitude and judgments of articulatory-phonetic differences will be discussed. [Work supported in part by NIH-NINCDS and the Louisiana Lions Eye Foundation.]

U9. The use of rate information during phonetic perception depends on pitch continuity. Kerry P. Green, Erica B. Stevens, and Patricia K. Kuhl (Department of Speech and Hearing Sciences, University of Washington, Seattle, WA 98195)

Studies have demonstrated that speaking rate provides an important context for the perception of certain acoustic properties of speech. For example, syllable duration, which varies as a function of speaking rate, has been shown to influence the perception of voice-onset-time (VOT) for syllable-initial stop consonants. The purpose of the present experiments was to examine the influence of syllable duration when the initial portion of the syllable was produced by one talker, and the remainder of the syllable was produced by a different talker. A short duration and a long duration /bi-pi/ continuum was synthesized with pitch and formant values appropriate to a female talker. When presented to listeners for identification, these stimuli demonstrated the typical effect of syllable duration on the voicing boundary: a shorter VOT boundary for the short stimuli relative to the long. An /i/ vowel, synthesized with pitch and formant values appropriate to a male talker, was added to the end of each of the short tokens producing a new "hybrid" continuum. Although the overall syllable durations of the hybrid stimuli equaled the original long stimuli, they produced a VOT boundary similar to the short stimuli. In a follow-up experiment, two new /i/ vowels were synthesized. One had a pitch appropriate to a female talker with formant values appropriate to a male talker, while the second had a pitch appropriate to a male talker and formants appropriate to a female talker. These vowels were used to create two new hybrid continua. The results of this second experiment indicated that continuity of pitch but not of formant structure appears to be the important factor in the integration of speaking rate within a syllable. [Work supported by NIH.]

U10. Identification of onset order for two-tone components: Implications for voicing perception. Jody K. Layer and Richard E. Pastore (Department of Psychology, SUNY University Center at Binghamton, Binghamton, NY 13901)

A general account of voicing perception for initial position stop consonants varying in VOT has been offered based on the identification of onset order of component events. Pisoni [*J. Acoust. Soc. Am.* 61, 1352-1361 (1977)] reported categorical perception for nonspeech tonal stimuli, although the exact boundary locations differed from VOT speech boundaries. The current research replicates and extends the Pisoni study. Labeling task boundaries were in agreement with Pisoni, and a two-interval, forced-choice (2IFC) task yielded boundaries even shorter than those for the labeling task. Most synthetic speech stimuli varying in VOT employ

an initial burst at onset. Adding a noise burst to the beginning of the tone stimulus shifted the temporal-order labeling boundaries to significantly longer onset asynchronies that are consistent with those reported for VOT. Implications for temporal-order processing and speech perception will be discussed. [Work supported by NSF.]

U11. Phonetic and nonphonetic fusion in duplex perception. Lynne C. Nygaard and Peter D. Eimas (Department of Cognitive and Linguistic Sciences, Brown University, Box 1978, Providence, RI 02912)

When listeners hear a synthetic consonant-vowel syllable split with its third-formant transition presented to one ear and the rest of the syllable (the base) presented to the other ear, they report two distinct percepts: A nonspeech chirp sound in the ear presented the transition and a complete syllable in the ear presented the base. The present studies investigated the nature of fusion of information yielding the phonetic percept. By introducing a third-formant transition into the base, fusion of information in the nonspeech percept was also studied. In this case, listeners reported hearing both a nonspeech chirp centrally lateralized and a complete, but now clearer, syllable in the ear presented the base. Experiment 1 replicated the original phenomenon of duplex perception and demonstrated reliable phonetic and nonphonetic fusion of the isolated and embedded transitions. In experiments 2-5, fundamental frequency, amplitude, spectral composition, and onset times of the chirp were varied relative to the base that did or did not contain a third-formant transition. It was found that listeners' ability to detect the additional third-formant transition differed for the phonetic and chirp percepts for each of the four variables. These results suggest that, as expected, the mechanisms for the perception of speech and location are different. Moreover, neither system captures all the available information and thus the fusion of information can occur in both directions.

U12. Articulatory and acoustic signatures of selected monosyllabic words. Ray D. Kent and Gary Weismer (Department of Communicative Disorders, University of Wisconsin-Madison, 1975 Willow Drive, Madison, WI 53706)

At previous meetings of this Society, acoustic signatures for selected words from an intelligibility test have been described. The signatures consist minimally of the formant trajectories of the syllable nucleus of each word. The formant frequency information can be augmented by other acoustic information, such as information on nasal murmurs, stop bursts, silences, and frication intervals. A major goal of this work has been to determine the interspeaker variance of the signatures and to evaluate their usefulness in characterizing speech disorders. This report extends the research to include articulatory data obtained by x-ray microbeam. The articulatory signatures take the form of planar motion paths of radiopaque markers attached to the articulators. Articulatory data were obtained from five talkers. Selected results will be presented to illustrate the articulatory signatures and to show their relationships to acoustic signatures of the same words. The primary goal of this report is to describe intraspeaker and interspeaker variances as they relate to the possibility of developing normative standards for articulatory-acoustic signatures. [Work supported in part by NIH.]

U13. Consonant types and tone in Siamese revisited. John Drury (DMLL, Cornell University, Ithaca, NY 14853)

This is a report of a production study of the effects of consonant types and lexical tones on F_0 before and after the release of the consonantal gesture in the speech of one speaker of Central Thai. The study is designed to determine whether the duration or magnitude of the effects of consonants on F_0 varies consistently with the lexical tone of the syllable in which the consonants occur. The main findings are that (1) segments that depress F_0 have a smaller effect on the F_0 of lower tones than of higher

tones, while (2) segments that elevate F_0 have approximately the same effect on F_0 for all tones; and (3) consonants have only a brief effect on post-release F_0 , as in previous study [J. Gandour, *J. Phon.* 2, 337–350 (1974)]. The findings are discussed in terms of a model in which the measurable F_0 of an utterance can be predicted by combining segmental F_0 /time functions with prosodic F_0 /time functions.

U14. Assessing the perception of foreign speech sounds. Bernard Rochet (Department of Romance Languages, University of Alberta, Edmonton, Alberta T6G 2E6, Canada)

This paper reports the results of a series of tests in which a set of synthetic /sVt/ stimuli was presented to native speakers of Standard French and Western Canadian English in order to assess the effect of phonemic interference as a perceptual phenomenon. Each linguistic group was asked to categorize the vowel portion of each stimulus in terms of its L_1 phonological system, and, in a second test, in terms of its L_2 system. In addition, subjects were asked to rate each stimulus as to the adequacy of its vowel as a member of its class. The results provide a representation of the subjects' perceptual categorizations of a subset of the vowel space of their native and their target languages (English /u/ and /U/, and French /u/ and /ü/). Analysis reveals that English /u/ is characterized by significantly higher F_2 values than its French counterpart, and that few subjects at the beginner or the intermediate level perceive L_2 phonemes in the same way as L_2 native speakers; for most, the acoustic specifications of L_2 vowels match closely those of their L_1 vowels, or are intermediate between the L_1 and L_2 values, in keeping with the notion of phonetic learning.

U15. Categorization of nonspeech stimuli. Xiao-Feng Li, Jody K. Layer, and Richard E. Pastore (Department of Psychology, SUNY—Binghamton, Binghamton, NY 13901)

A number of studies have claimed that a continuum consisting solely of auditory chirps or bleats produces uniform chance labeling and discrimination performance. However, when these auditory continua are added to an appropriate syllable base, they produce a categorically perceived place continuum. The current study demonstrates that practice allows subjects to categorically perceive these auditory continua in a manner equivalent to that reported for speech place continua. Three sets of 16 sinusoidal stimuli were synthesized with 40-ms initial transitions that were analogous to the second formants of syllables. The starting frequency of the transitions varied in 15 equal steps from 1140 to 2420 Hz. The duration of the final steady-state portion of the stimuli was 190, 40, or 0 ms [bleats, short bleats, and chirps] with total duration of 230, 80, or 40 ms, respectively. The study consisted of 4 practice days with initial testing on each day, and followed by 2 days of reevaluation of labeling performance. [Work supported by NSF.]

U16. The role of spectral, temporal, and dynamic cues in the perception of English vowels by native and nonnative speakers. Salvatore Miranda (Department of Linguistics, University of Connecticut, U-145, Storrs, CT 06269-1145) and Winifred Strange (Departments of Communication Sciences and Disorders and Psychology, University of South Florida, Tampa, FL 33620)

Previous cross-language studies have demonstrated that different languages may utilize different phonetic features to signal phonemic distinctions. This study investigated which phonetic features are utilized by language learners perceiving speech in the target language and if the acoustic parameters that specify the features are processed in the same way by native and nonnative speakers. Native speakers of English and students of English as a second language (ESL) identified American English vowels that were electronically modified to vary systematically in the type of acoustic information available. The role of three types of acoustic infor-

mation was investigated: vowel targets available in quasisteady-state syllable nuclei; dynamic information contained in rapidly changing formant transitions; and intrinsic duration information. Results indicated a remarkably similar error pattern across conditions for both native and nonnative subjects, except for one less-advanced subgroup of nonnatives who identified vowels in certain modified conditions more poorly than vowels in an unmodified control condition, suggesting that less-advanced nonnatives may need all possible sources of information to identify English vowels correctly. [Work supported by NINCDS.]

U17. Perception of speaker age using a paired stimuli technique. Richard J. Morris (Department of Communicative Disorders, The Florida State University, Tallahassee, FL 32306-2007) and W. S. Brown, Jr. (University of Florida, Gainesville, FL 32601)

Listeners have consistently demonstrated the ability to estimate the age of taped voice within a few years of the chronological age of the speaker. In this study, a paired comparison technique was used to determine the consistency of raters' judgments. The listeners rated the age of voices from two age groups of speakers. Both intrajudge and interjudge consistency were determined. The speakers for the study consisted of ten women who were 20–30 years of age and ten women who were 75–90 years of age. A 3-s sustained /a/ at comfortable frequency and intensity levels was used as the stimulus for the listeners. The listeners were 22 undergraduate students who rated the second voice in each pair relative to the first one. The voices were rated as sounding "much older," "older," "the same age," "younger," or "much younger" than the first voice. A multiple correlation statistical paradigm was used to analyze the data. The results will be discussed in relation to theories of aging speech production and perception.

U18. Source characteristics: A study of hand clapping. Stuart A. Tousman, Richard E. Pastore, and Steven Schwartz (Department of Psychology, SUNY—Binghamton, Binghamton, NY 13901)

The present research evaluated the capabilities of listeners to determine source characteristics from acoustic signal properties. Repp [*J. Acoust. Soc. Am.* 81, 1100–1110 (1987)] demonstrated that naive listeners have some ability to determine the source characteristic of hand position given the acoustic signal of a clap. The purpose of this hand clapping experiment was (1) to more directly assess the capabilities of naive listeners to determine source characteristics, (2) to determine the physical aspects of the acoustic signal that are highly correlated with the source, and (3) to train subjects so that performance could be improved. The production phase of the experiment consisted of recording claps in five different hand positions. The perceptual phase consisted of a within clapper same/different task with listeners asked to judge whether the hand position was the same (never physically identical) or different. Half of the listeners tested received feedback. Preliminary results indicate that naive listeners could distinguish hand position from only a few clappers. Furthermore, training over 360 trials was only marginally beneficial in improving performance. Results will be discussed in terms of different methods of assessing and improving listener performance. [Work supported by BRSG program grant.]

U19. Perceptual units of the infant cry. Taeko Tsukamoto (Konan Women's College, Morikita-machi, Higashinada-ku, Kobe, 658 Japan) and Yoh'ichi Tohkura (ATR Auditory and Visual Perception Research Laboratories, Inuidani, Seika-cho, Kyoto, 619-02 Japan)

Perceptual units for category identification of infant cries have been studied. Three cry categories discussed in this paper are the hunger cry, the call cry (i.e., cry calling for infant–mother interaction), and the anger cry. The original samples have been classified into these three categories. In order to generate stimuli to be used in the perceptual experiments, each

of the cry samples is first segmented into single-segment units according to breath groups of the cry samples. Next, the single-segment units are combined with each other in temporal order to generate two-, three-, five-, and seven-segment unit stimuli. In addition to the multisegment unit stimuli thus obtained, the single-segment units and the three original samples are used as one-segment unit stimuli and full-segment unit stimuli, respectively, in the perceptual experiments. In identifying the cry stimuli, subjects are instructed to make a forced choice among the three cry categories. The experimental results show that category identification rates are greatly dependent upon the number of segments making up each stimulus. However, the identification rates are temporarily saturated at two-segment units in the call cry and at three- to five-segment units in the hunger and anger cries. This fact indicates that the units with two to five segments are the perceptual units. Temporal duration of the perceptual units across all three categories is similar (i.e., about 6–8 s).

U20. Respiratory response to oral flow and pressure perturbations during speech. Anne Putnam Rochet (Department of Speech Pathology and Audiology, University of Alberta, Edmonton, Alberta T6G 0T2, Canada), Kathleen E. Morr (Department of Dental Hygiene, Idaho State University, Pocatello, ID 83209), and Donald W. Warren (Dental Research Center, University of North Carolina, Chapel Hill, NC 27514)

Evidence for the regulation/control hypothesis of Warren for speech aeromechanics [Cleft Palate J. 23, 251–260 (1986)] was observed in nine normal men who produced /p/ and /s/ under conditions in which a translabial device released intraoral pressure during /p/, and bite blocks opened the anterior bite artificially during /s/. Intraoral pressures, oral flows, and respiratory volumes associated with utterance were the dependent variables. For /p/, intraoral pressure decreased and translabial air leakage increased as bleed orifice area increased, but mean peak intraoral pressure never fell below 4 cm H₂O. For /s/, mean oral flow increased slightly as the anterior bite was opened, but mean peak oral pressure did not vary significantly from control values. Mean flow on the postconsonantal vowels did not vary significantly across the experimental conditions. There were no significant differences among mean inspiratory volumes within or across translabial bleed conditions, nor within or across bite-block conditions. Expiratory volume increased significantly as bleed orifice area increased, but not as bite-block size increased. These data may reflect the regulation control options inherent in each condition. The respiratory system alone could be relied upon to maintain intraoral pressure within requisite limits for speech during the bleed conditions; during the bite-block trials, speakers could employ increased respiratory drive, structural adjustments of the sibilant orifice, or both, to compensate for aeromechanical perturbation introduced by the blocks. [Work supported by AHFMR.]

U21. Accuracy and reliability of formant frequency measurements in synthetic infant CV syllables. Nancye C. Roussel, Raymond G. Daniloff, and Creighton J. Miller (163 Music and Dramatic Arts Building, Louisiana State University, Baton Rouge, LA 70803-2606)

An investigation of the reliability and accuracy of measurement of infant formant frequencies was conducted. Four factors known to render formant measurement difficult—pitch inflection, glottal noise, jitter, and nasalization—were incorporated into synthesized CV (C = /b,d/, V = /i,a,e/) syllables appropriate to 3-month-olds. The syllables were of high intelligibility and quality. Three fundamental frequency contours (standard rise/fall, rising, flat), three F₀ levels (325, 375, 425 Hz), four voice/noise ratios (+20, +5, 0, -15), three frequency jitter rates (250, 500, 750), and one degree of nasal coupling were used. Formant frequency measurements were made in two ways for each token: (1) as single FFT cross sections using a 10-ms window and (2) as averaged, time-advanced FFT spectra. In the second condition, three successive FFTs taken at 2.5-ms rightward shifts of the 10-ms sampling window were averaged to produce the FFT spectrum for measurement. Three observers independently measured F₁, F₂, and F₃ and results were compared to synthesis parameter values. Inter- and intraobserver reliability was also computed. Results are discussed with regard to the influence of consonant place and vowel type upon accuracy in each of the four conditions of waveform perturbation.

U22. Accuracy and reliability of formant frequency measurements in infant CV-like utterances. Raymond G. Daniloff, Nancye C. Roussel, and Creighton J. Miller (163 Music and Dramatic Arts Building, Louisiana State University, Baton Rouge, LA 70803-2606)

The accuracy and reliability of measurements of infant formant frequencies were investigated. A total of 40 CV-type tokens (C = labial- or velarlike stop clesant, V = /i/-like or /e/-like vocalic segment) were selected from recordings of vocalizations of two infants, ages 2–4 months. The tokens were of good acoustic quality and were taken from recordings of postfeeding mother–child interactions. Formant frequency measurements were made from both single FFT cross sections using a 10-ms sampling window and averaged, time-advanced FFT spectra. Time-advanced FFTs were computed using three successive FFTs taken at 2.5-ms rightward shifts of the 10-ms window. Three investigators independently measured formant frequencies in all tokens. Wideband 500-Hz spectrograms of each token were used to establish baseline values of formant frequencies for comparison with measured FFT values. Inter- and intraobserver reliability was also determined.

U23. Fundamental and formant frequencies in an adolescent population. Susan R. Merino and Terrance M. Nearey (Department of Linguistics, University of Alberta, Edmonton, Alberta T6G 2E7, Canada)

Although measurements of children's, adult male's, and female's vowels are readily available in the literature, few studies can be found that deal with the resonances of the adolescent vocal tract. The adolescent population is of interest because of the variation in vocal tract lengths that it provides and the consequent effects of this variation on the formant frequencies of the vowels. Recordings were made of 11 vowels produced by 30 junior high school students, 16 male and 14 female, between 11 and 16 years of age. The students were attending the same school and were all native speakers of the same dialect. Measurements of the fundamental and formant frequencies were taken from two tokens of each 11 vowels for all of the speakers. A preliminary analysis of the data will be presented.

U24. The application of frequency-domain adaptive filters to speech enhancement. Tarek I. Haweel and Peter M. Clarkson (Department of Electrical and Computer Engineering, Illinois Institute of Technology, Chicago, IL 60616)

Several authors have applied the time-domain least-mean-squares adaptive filter to the problem of (voiced) speech enhancement. Generally, these efforts have achieved only limited success due, in part at least, to the nonuniform convergence of the adaptive filter when faced with frequency components of highly disparate spectral power (the so-called "eigenvalue disparity" problem). This problem is addressed by employing the normalization capacity of the frequency-domain adaptive filter (FDAF). The first part of this paper deals with the analysis of the FDAF for strictly harmonic signals (this reflects the quasiperiodic nature of voiced speech). It is shown that the behavior of the filter for each weight of the FDAF can be described by a linear transfer function relating the desired input to the output signal. It is further shown that the product of the input power and the feedback component (in each frequency bin) determines the stability and convergence of the filter. Normalization can be achieved by adjusting this product for each weight. Simulations comparing time- and frequency-domain approaches have shown that the FDAF enhancer significantly improves performance.

U25. The intelligibility of digitally high-pass filtering the discrete digital frequencies in amplitude- and time-quantized speech. Edward M. O'Brien (Bioengineering Program, Texas A&M University, College Station, TX 77843-3120)

Speech was first amplitude quantized into two levels, depending on whether the speech signal was above or below a 0-V reference. Time quan-

tization was next accomplished using clocking rates of 25, 16.7, 10, and 6.25 kHz. Thus only discrete digital frequencies remain in the processed waveform. Having the speech in a digital format makes it amiable for further processing. Determining the discrete digital frequencies is relatively straightforward and these can then be displayed onto the skin using electrical stimulation to implement a tactile hearing aid. Since only a finite number of electrodes can be used, the effect of high-pass filtering the quantized speech was investigated using seven normal-hearing subjects. Phonetically balanced monosyllabic words were used as the test tokens. With no high-pass filtering, the percent word intelligibilities were 91%, 86%, 77%, and 54% for quantizing rates of 25, 16.7, 10, and 6.25 kHz, respectively. Several filters were tested for each quantizing rate. However, 70% word intelligibility (considered adequate for conversational speech) was obtained with only 15, 11, and 9 discrete digital frequencies remaining in the filtered speech for quantizing rates of 25, 16.7, and 10 kHz, respectively.

U26. Voice quality in males: A multidimensional scaling study. Marylou Pausewang Gelfer (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

The purpose of this study was to investigate the perceptual dimensions that listeners use in discriminating male voices, and how these dimensions relate to perceptual rating scale judgments. Results were compared to earlier research using female voices. Listeners were 18 certified speech pathologists who judged the dissimilarity of sentences produced by 20 males (20–26 years of age). The listeners also evaluated each voice on 17 perceptual rating scales (e.g., high pitch–low pitch, clear–hoarse). Results of the multidimensional scaling (MDS) analysis produced a four-dimensional solution. A stepwise linear regression related the perceptual attributes of speech rate and vocal pitch to the first two dimensions, while the second two dimensions were best explained by descriptors pertaining to quality (e.g., pleasant, smooth, melodious). Compared to the MDS analysis of female voices, the variance accounted for was reduced, suggesting more random variability among male voices, or that these voices were more difficult for listeners to rate in a consistent manner. Combined results from both investigations suggest that the perceptual dimensions that listeners use in discriminating among male voices may not be substantially different from the dimension used in discriminating among female voices. [Research supported by BRS Grant S07 RR07031, DRR, NIH.]

U27. Variations in measured voice quality related to linguistic and nonlinguistic factors. Gerald W. McRoberts (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511 and University of Connecticut, Storrs, CT 06268) and Alice Faber (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

Voice quality (VQ) in English has generally been treated as a suprasegmental, not varying across individual speech segments. Ladefoged [in *Vocal Fold Physiology*, edited by Bless and Abbs (1983)] has measured the spectral correlates of VQ differences in languages in which both VQ and tone (F_0) are phonemically contrastive. Since VQ variation is not contrastive in English, it is in theory available to enhance other linguistic contrasts, or may vary as a function of nonlinguistic factors, such as fatigue or affect. Following Ladefoged, VQ was measured as the difference between the amplitude of F_0 and the amplitude of the most intense harmonic in F_1 using discrete Fourier transform spectra. Two separate corpora of utterances by subjects without known voice pathology were analyzed. One corpus compared qualitatively similar stressed and unstressed vowels (Vs) produced by a single speaker in "fresh" and "fatigued" circumstances. The other compared utterances produced by two trained actors in contrastively and normally stressed conditions, and simulated positive and negative affective conditions. These analyses suggest that this measure of VQ differentiates between Vs in (1) stressed and unstressed conditions, and (2) normally and contrastively stressed conditions. However, it did not differentiate between Vs in fresh and fatigued conditions or in the simulated affective conditions. The results are discussed in terms of

the use of VQ and its relation to F_0 for conveying linguistic and nonlinguistic distinctions. [Work supported by NIH Grants NS 07237, HD 1994, NS 24655.]

U28. Intelligibility of speech produced in noise and while wearing an oxygen mask. Z. S. Bond (Department of Linguistics, Ohio University, Athens, OH 45701) and Thomas J. Moore (Armstrong Aerospace Medical Research Laboratory, Wright-Patterson AFB, OH 45433)

A number of laboratories have described relatively systematic changes in the acoustic-phonetic structure of speech produced in the presence of noise relative to that produced under more benign speaking circumstances. At least some of these changes may be attempts to make speech clearer and more intelligible. In order to evaluate the intelligibility of speech produced in various circumstances, a male talker speaking isolated words in quiet and in the presence of 95 dB SPL of pink noise was recorded. Speech samples were collected both with and without an oxygen mask. The peak amplitudes of each word were normalized, and the words were mixed with pink noise and presented for identification to 45 native and to 10 nonnative English speakers. The order of relative intelligibility was the same for both groups. Speech produced while listening to noise alone was most intelligible. Speech in the laboratory and in mask conditions was considerably less intelligible. The noise + mask condition led to marginally the least intelligible speech. [Work supported by AFSOR.]

U29. Speech intelligibility as a function of environment and speaking style. Karen L. Payton, Rosalie M. Uchanski, and Louis D. Braida (R.L.E. 36-747 MIT, 50 Vassar Street, Cambridge, MA 02139)

Sentences spoken "clearly" have been shown to be more intelligible than those spoken "conversationally," both for hearing-impaired listeners in a quiet background [Picheny *et al.*, *J. Speech Hear. Res.* 28, 96–103 (1985)] and for normal-hearing listeners in a noise background [R. M. Uchanski, Ph.D thesis, MIT (1988)]. This study examined the effects of noise and reverberation on the intelligibility of these two speaking styles for listeners with normal hearing. Six environmental conditions were tested: three reverberation times (anechoic, $T = 0.18$ s, and $T = 0.6$ s) by two levels of noise ($+\infty$ and 0 dB S/N). As the environmental degradation increased (calculated from the STI), intelligibility scores decreased for both speaking styles. However, clear speech was more intelligible than conversational speech for each condition. Further, the clear speech scores declined at a more gradual rate than the conversational scores as the STI decreased. Envelope spectra of clear and conversational speech are being examined to derive a predictor of intelligibility that includes the characteristics of both the speech source and the acoustic environment.

U30. Speaker identity verification of spectrally ambiguous and distinguishable speech. Robert J. Logan (Department of Psychology, SUNY—Binghamton, Binghamton, NY 13901), Timothy C. Feustel, George A. Velius (Bell Communications Research, 435 South Street, MRE 2E 236, Morristown, NJ 07960), and Richard E. Pastore (Department of Psychology, SUNY—Binghamton, Binghamton, NY 13901)

Feustel *et al.* [*J. Acoust. Soc. Am. Suppl.* 1 83, S55 (1988)] evaluated performance of humans in a speaker identity verification (SIV) task with the intent of improving the performance of a LPC-cepstral based verification algorithm. This study uses the general methodology and signal processing techniques developed by Feustel *et al.* to further investigate the nature of potential invariant acoustic information (within a speaker) used by humans to verify a speaker's identity. Subjects were presented natural speech or modified natural speech stimuli drawn from a corpus of spectrally distinguishable and spectrally ambiguous utterances. The results suggest that humans can use information from both the glottal (source) waveform and spectral envelope. Overall, the glottal waveform provides more reliable information. The results will be discussed in this context.

U31. Speech recognition using codebooks trained from a synthesized seed. L. A. Smith, B. L. Scott, L. S. Lin, and J. M. Newell (Scott Instruments Corporation, 1111 Willow Springs Drive, Denton, TX 76205)

Speech sounds generated by a simple waveform synthesizer were used to create a 16-vector codebook for vector quantization (vq). This codebook was used as a seed for training over the TI-20 isolated word database, which was then used to test the resulting codebooks using speech recognition. The database was split into two sets of speakers and two codebooks were created for each set, one trained on one pass over the speech and the other trained iteratively. All codebooks were trained adaptively, resulting in codebooks of 31 vectors—16 high-energy and 15 low-energy vectors. Recognition was performed using a conventional DTW algorithm. The baseline recognition accuracy (using no vq) was 97%; the accuracy for the 16-vector synthesized codebook was 91.8%. Accuracy using trained codebooks ranged from 94% (two-sided vq) to 95.1% (one-sided vq with a normalized match distance). Crossing speaker sets (resulting in speaker-independent codebooks) yielded 94.9% recognition accuracy. The results indicate that speech synthesis can be used to decrease the training needed to develop multispeaker and speaker-independent codebooks.

U32. Improvements to a speaker-independent algorithm. Brian Scott, Lisan Lin, Mark Newell, and Lloyd Smith (Scott Instruments Corporation, 1111 Willow Springs Drive, Denton, TX 76205)

In May 1988, a speaker-independent recognition algorithm was described at the 115th Meeting of the Acoustical Society of America [Scott *et al.*, J. Acoust. Soc. Am. Suppl. 1 **83**, S55 (1988)]. The algorithm yielded 95.4% recognition accuracy on the 20-word TI database obtained from the National Bureau of Standards. The system has since been adapted for use over the telephone. In so doing, a new database was developed consisting of 16 words (zero-nine, oh, yes, no, cancel, terminate, and help) as spoken by 11 males and 11 females from various locations across the country. Although small (1001 utterances), the database represented a significant challenge as compared to the one obtained from the NBS. There were fewer training passes per word, more speakers, and there was considerably more noise in the database. Initial tests on this database yielded accuracies of approximately 60%. Four major enhancements to the algorithm improved the accuracy on this database to 95.3%. Two of the enhancements compensate for time alignment problems inherent in both linear and nonlinear time normalization routines through the use of prosody and redundancy. The other two enhancements include the extraction of additional features and better use of variances. Results indicate that acceptable speaker-independent recognition can be obtained with minimal training and processor requirements, given effective normalization routines during the front-end signal processing.

U33. APS: An environment for acoustic phonetic research. Gordon S. Watson (Centre for Speech Technology Research and Department of Artificial Intelligence, University of Edinburgh, 80 South Bridge, Edinburgh EH1 1HN, Scotland)

Continuous speech recognition systems based on acoustic phonetic features demand characterization of the relationship between phonetic segments and their acoustic values. This paper describes a software environment for exploring this relationship. The software system is known as APS—acoustic phonetic research with S (the standard UNIX statistical package). APS implements a language formalism for describing segments of speech, forming an interface between the domain of phonetics and a rich speech database. APS enables the automatic retrieval of all tokens of a given segment type together with their acoustic parameter values in a phonemic context specified by the user. For example, using APS, it is possible in a single instruction to perform statistical operations (e.g., mean, variance) on the durations of all word-final oral stops following stressed vowels [T. H. Crystal and A. S. House, J. Acoust. Soc. Am. **83**, 1553–1573 (1988)]. The motivation for APS and the principles of its design are described and examples of its usage are given.

U34. An algorithm for distinguishing between voiceless stops and voiceless fricatives. LaDeana F. Weigelt, Steven J. Sadoff, and James D. Miller (Central Institute for the Deaf, 818 South Euclid, St. Louis, MO 63110)

An algorithm has been developed for the purpose of voiceless stop/fricative distinction in word-initial position using three measures of the waveform: zero crossings, energy, and derivative of the energy over time (termed rate of rise, ROR). The ROR is used as the primary classifier while energy and zero crossings are used to discard spurious, irrelevant peaks. Peaks in ROR are considered in order of magnitude for relevance to the stop/fricative distinction. The resulting algorithm was tested on 720 CVC tokens (four male speakers, four female speakers, three stops and an affricate [P,T,K,CH], five fricatives [S,SH,F,TH,H], and ten vowel contexts [IY,IH,EH,AE,AA,AH,AO,UH,UW,ER]). Data from two male and two female speakers (360 tokens) were used as a training set, and the remaining data (360 tokens) were used as a test set. The overall success rate was 96.8%. [Work supported by AFOSR.]

U35. Pattern processing of stop consonants: Part II. L. Garrison-Shaffer (Division of Humanities and Social Sciences, Penn State University—The Behrend College, Station Road, Erie, PA 16563)

The roles of F_1 , F_2 , and vowel environment in stop consonant pattern processing were investigated using interaural transfer and selective adaptation. Previous results indicate that different patterns may exist for processing of stops, depending upon whether the stop occurs in a front or back vowel environment [L. Garrison, J. Acoust. Soc. Am. Suppl. 1 **83**, S70 (1988)]. Such pattern processing may reflect differences in the placement of F_2 relative to F_1 . This experiment systematically manipulated the relationships between formants as well as vowel environment for CV syllables. These manipulations resulted in six stimuli that were used as adaptors on a [ba]–[da] test series. One adaptor shared its vowel percept ([ba]) and three other adaptors shared vowel category (back) with the test series. Finally, two adaptors were from a front vowel category. The results will be discussed in relation to how formant relationships and vowel environment affect pattern processing of stop consonants. [Work supported by NINCDS.]

U36. Acoustic features of nasal consonants in VC contexts. Yingyong Qi and Robert Allen Fox (Division of Speech and Hearing Science, The Ohio State University, 324 Derby Hall, 154 North Oval Mall, Columbus, OH 43210-1372)

The acoustic features of the nasals [m,n] in CV syllables and [m,n,ŋ] in VC syllables in English were first analyzed and compared using cepstrally smoothed running FFT spectra. Clear differences between nasals in CV and VC syllables were obtained. For example, the spectral energy transitions from vowel to nasal in VC syllables were found to be much less dramatic than in CV transitions. Next, given the recent interest in the efficacy of auditory representations in speech recognition schemes, the nasals in VC contexts were examined in terms of auditory transformed running spectra. Several features of interest were obtained: The spectra were generally dominated by the second formant and, when the preceding vowel was a low vowel, nasal place of articulation was distinguished by the second formant transition, which converged to 16–17 ERB for [m] and 19–20 ERB for [n]; [ŋ] was characterized by little formant movement. Consistent nasal place features were not found in the context of the vowels [i] and [u]. Finally, since the antiresonances of a nasal may provide place of articulation information, the system zeros of the nasals were analyzed using parametric spectral analysis based on the autoregressive moving average (ARMA) process. Results indicated that the antiresonances of nasals in a VC syllable could be consistently estimated by a two-step AR approximation method that could be used to distinguish reliably between [m] and [n].

U37. ESPRIT: A signal processing environment with a visual programming interface. Robert T. Gayvert, John A. Biles, Harvey Rhody (RIT Research Corporation and Rochester Institute of Technology, 75 Highpower Road, Rochester, NY 14623), and James Hillenbrand (Speech Pathology and Audiology, Western Michigan University, Kalamazoo, MI 49008)

ESPRIT (Explorer speech processing system from the Rochester Institute of Technology) is an integrated speech research development environment that runs on the TI Explorer, optionally augmented by the TMS-320 based Odyssey DSP board. The goal of ESPRIT is to provide speech scientists, linguists, and engineers an intuitive environment in which to collect, process, and display speech signals. ESPRIT's module editor allows users who are not programmers to draw data-flow programs made up of built-in and user-defined speech processing algorithms, display functions, and standard utilities. ESPRIT's display editor allows users to manipulate the graphical displays that result from running these programs to zoom, scroll, rearrange, take precise measurements, and perform a variety of other operations. While ESPRIT provides standard signal processing algorithms (FFT, LPC) and displays (waveforms, spectrograms, waterfalls, spectral slices), users who develop their own Lisp or TMS 320 programs can easily install them to replace or augment the standard software. [Work supported by Rome Air Development Center under Contract No. F30602-87-D-0090.]

U38. Effects of tree structure and statistical methods on broad phonetic classification. James W. Delmege (RIT Research Corporation and Rochester Institute of Technology, 75 Highpower Road, Rochester, NY 14623), James Hillenbrand (Speech Pathology and Audiology, Western Michigan University, Kalamazoo, MI 49008), and Robert T. Gayvert (RIT Research Corporation and Rochester Institute of Technology, 75 Highpower Road, Rochester, NY 14623)

The goal of this project was to develop a system for assigning individual frames of a speech signal to one of four broad phonetic categories: vowel-like, strong fricative, weak fricative, and silence. Classification results were compared from a *K*-means clustering algorithm and a maximum likelihood distance measure. In addition to the comparison of statistical methods, this study compared classification performance using several tree-structured, decision-making techniques. Training and test data consisted of various combinations of 98 utterances produced by five male and five female speakers. Results showed very little difference between the *K*-means and maximum likelihood methods. However, the nature of the decision tree had a significant effect on the performance of the classifier. [Work supported by Rome Air Development Center and the Air Force Office of Scientific Research as part of the Northeast Artificial Intelligence Consortium (Contract No. F3060285-C-0008) and by Redcom Laboratories, Victor, NY.]

U39. Microcomputer system for analysis of the verbal behavior of patients with neurological and laryngeal diseases. Maurice Yunik, Boyan Boyanov, and Budi Rahardjo (Department of Electrical Engineering, University of Manitoba, Winnipeg, Manitoba R3T 2N2, Canada)

The microcomputer system for verbal behavior analysis is built on the basis of the IBM-PC. It consists of two parts: hardware and software. The hardware is the condenser microphone and a two-channel ADC board, including: amplifiers with adjustable gain; two Bessel analog filters with cutoff frequencies of 800 Hz and 6 kHz; and two ADCs allowing sampling rate up to 40 kHz. The software is a program package which allows high precision pitch detection by means of the autocorrelation function; finding the separate pitch periods by means of amplitude selection and parabolic interpolation; pitch period perturbation; number of subharmonic-to-fundamental frequency power; separation of segments where the signal is strictly periodic; realization of spectral analysis over these segments by means of fast Fourier and fast Walsh transforms; and evaluation of the degree of hoarseness from the spectra. Preliminary experimental research

shows that the system may be very useful for analysis of the speech of patients with neurological and laryngeal diseases as well as the analysis for musical sounds. Results from analysis of pathologic voices agree with those of Kasuya *et al.* ["Preliminary Experiments on Voice Screening," in *Proceedings of the Symposium on Voice Acoustics and Dysphonia*, Gotland, Sweden (August 1985)] and Laver *et al.* ["An Acoustic Screening System for the Detection of Laryngeal Pathology," *J. Phon.* **14**, 517-524 (1986)].

U40. Effects of changes in stress and in rate of speech on vowels. Marios Fourakis (Central Institute for the Deaf, 818 South Euclid, St. Louis, MO 63110)

The effects of changes in stress and in rate of speech on the acoustic characteristics of American vowels as produced by native speakers in real words were examined. The words were embedded in a carrier sentence. Subjects were instructed to produce four repetitions of the carrier sentence for each test word under each of four conditions: (a) slow, (b) fast, (c) stress on the test word, and (d) stress on an attached dummy syllable. There were 32 tokens per vowel for each speaker. The utterances of one male speaker have been completely analyzed and plotted in an auditory-perceptual space [J. D. Miller, "Auditory-perceptual interpretation of the vowel," *J. Acoust. Soc. Am.* (in press)]. The vowel target zones described by Miller correctly classify 85% of the vowel data, regardless of stress and rate conditions; and with minor modifications, the accuracy of classification is improved. Data for one female speaker will also be presented. Furthermore, measurements of sentence, word, and isolated vowel durations will be presented for both speakers. [Work supported by NINCDS.]

U41. Speaker-independent word recognition with Lombard speech. Brian A. Hanson, Ted H. Applebaum, and Gregory R. De Haan (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

To investigate the robustness of a small vocabulary, speaker-independent word recognizer trained on normal speech, testing was done using Lombard speech. The Lombard database was collected from talkers exposed to 85 dB SPL white Gaussian noise played through headphones. Recognition tests utilized a hidden Markov model recognizer with separate codebooks for static and dynamic cepstral features [Gupta *et al.*, in *Proc. IEEE ICASSP-87*, 697-700 (1987)]. The front-end analysis used either standard linear prediction (LP) or a version of perceptually based linear prediction. (This PLP has filtering functions with 3-dB bandwidths of 1 Bark, a correction to Eq. (3) in Hermansky *et al.* [*Proc. IEEE ICASSP-85*, 509-512 (1985)].) The effects of the analysis technique, analysis order, distance measure (cepstral or weighted cepstral), and spectral features (static, dynamic, or integrated) were studied. The utility of static and dynamic features differed greatly according to the analysis method and distance measure. For example, for low (5th)-order PLP, the dynamic features alone performed much better than either static or integrated features. Alternatively, for high (14th)-order LPC with weighted cepstral distance, the dynamical features alone performed much worse than either static or integrated features, yet contributed to improved recognition rates with integrated features.

U42. Acoustic-phonetic analysis of stress-induced speech. John W. Gordon (Boeing Aerospace Company, P. O. Box 3999, Mail Stop 82-58, Seattle, WA 98124)

The Lincoln Laboratories stress speech database (LLStyle) has been used to test various speech recognition algorithms that might be used in conditions such as stressful environments, under which the input speech can be expected to vary from its trained norm. To facilitate the correlation of performance degradations with speech variabilities, the entire LLStyle database (consisting of over 10 000 utterances) was transcribed. The pro-

gram SEARCH was then used to gather statistics of various acoustic-phonetic parameters. Grouping these statistics according to speaking style has provided some insight into which parameter changes cause problems in general for machine recognition. The analysis of individual words has helped to explain why certain assumptions made by some recognizer strategies (for constraining the search space) are invalid when applied to variable speech.

U43. Recognition of prosodic contrasts in literal and idiomatic utterances by native and nonnative speakers of English. Diana Van Lancker (Department of Neuroscience, University of North Dakota Medical School, Fargo, ND 58102)

Abilities of three groups—native English speakers, fluent nonnative English speakers, and advanced students of English as a second language (ESL)—to identify literal (L) and idiomatic (I) meanings of ambiguous sentences (e.g., "He was at the end of his rope") were tested. Tape-recorded utterances produced by two speakers with either intended L or I meanings were presented singly and in contrastive pairs; listeners judged whether each stimulus expressed an L or an I meaning. Native speakers performed significantly better than fluent nonnative speakers on both single and paired presentations, while the ESL speakers performed at chance on both tasks. Linguistic training was associated with greater accuracies in the native but not in the fluent nonnative speakers. Acoustic analyses had indicated that L and I utterances differed significantly on measures of fundamental frequency (F_0), number of pitch contours per utterance, lexical and overall utterance durations, and pausing. I utterances were higher in F_0 , and had fewer contours, shorter durations, and fewer pauses than the L utterances. The results replicate the original finding that native speakers can discriminate L from I meanings from prosodic cues alone [Van Lancker *et al.*, *J. Speech Hear. Res.* **24** (1981)], and further indicate that performance on the test was not attributable to a listening strategy and that native competency significantly affects abilities to process these prosodic cues.

U44. Digital processing of laryngeal images: A preliminary report. Raymond H. Colton, Janina K. Casper, David W. Brewer (Department of Otolaryngology and Communication Sciences, SUNY Health Science Center at Syracuse, Syracuse, NY 13210), and Edward G. Conture (Department of Communication Disorders, Syracuse University, Syracuse, NY 13210)

A low-cost microcomputer (Amiga 1000) and software were used to digitize and analyze slides, high-speed films, and videotapes of laryngeal images. The images were digitized with a 320 horizontal \times 200 vertical pixel resolution and at 16 grey levels. Various algorithms were used to analyze the images, which included color equalization, color merging, edge detection, and histogram analysis. An example is presented where the area of a nodule and the area of the glottis are measured using some of the processing techniques and compared to similar measurements obtained by hand. The quality of the digitized images obtained with this low-cost system is very good. Simple image processing techniques, commercially available, are well suited to the analysis of laryngeal images and should permit more sophisticated processing of movement of structures within the larynx.

U45. A comparison of x-ray microbeam and ultrasound measurements of the tongue surface during speech. Maureen Stone (Department of Rehabilitation Medicine, Room 6S235, Building 10, National Institutes of Health, Bethesda, MD 20892)

Tracking points on the tongue surface provides unique information about its movement, such as surface expansion/contraction, differences in rate of movement along the tongue's surface, and X - Y tongue position. Scanned images of the tongue surface provide different but equally unique information, such as posterior and anterior tongue relationships, tongue

shape, tongue rotation, and asymmetry. The present study used two quite different techniques (x-ray microbeam and ultrasound) in a comparative study of tongue movement. A single subject repeated /VCVC/ combinations of /s/, /l/, /i/, /a/, and /o/. The phonemic effects of the consonants on tongue shape were examined, as were the allophonic effects of the three vowels. Tongue minus jaw movements (ultrasound) were compared to tongue plus jaw movements (XRMB). The data indicated that tongue rotation, which was quite common, occurred about a fulcrum whose locus was shifted or obscured by jaw movement. Other results indicated that there was considerable expansion/contraction of the tongue surface during speech. In addition, the surface of the tongue moved at nonuniform rates that were both physiologically and phonemically based.

U46. Relationship of recorded EMG signals to within- and cross-utterance acoustic variation. Alice Faber (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511) and Lawrence J. Raphael (Herbert H. Lehman College, Bronx, NY 10468, The Graduate School, City University of New York, New York, NY 10021, and Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

Analysis of simultaneous acoustic and EMG recordings of one speaker of New York English producing /əpVp/ utterances under several conditions shows differences between inter- and intrautterance interactions. For three anterior genioglossus insertions (#1–#3) and one posterior insertion (#4), the peak activation amplitude relevant to articulation of the vowel was measured. Amplitude differences reliably differentiated target vowels. For vowel/muscle pairs with significant amplitudes, activity duration and phasing of the peak within the activity were also measured. Despite strong correlations among the peak amplitudes for the anterior insertions, these measures differed in their correlations with measures of F_0 , F_1 , and F_2 , both within and across utterances. Overall, F_2 correlated best with #1/duration, F_1 with #2/amplitude, and F_0 with #2/amplitude. For /i/, intertoken F_2 variation correlated best with #3/phasing, F_1 with #2/amplitude, and F_0 with #3/phasing. The posterior #4/duration correlated best with /uw/ F_1 and F_0 . These results suggest both the importance of applying similar intertoken analysis techniques to larger data sets and the advances in understanding of subglottal articulatory control likely to result. [Work supported by NIH.]

U47. Volume determination using acoustic resonance. Neil S. Rothman (The Johns Hopkins Applied Physics Laboratory, Laurel, MD 20810)

Cavities of irregular geometry can be modeled as Helmholtz resonators provided they adhere to specific dimensional limits. If the characteristics of the entrance region or "neck" of a cavity are held fixed, its resonant frequency will be a function only of its volume. This property of a Helmholtz resonator enables the determination of the volume of various cavities through resonant frequency measurements. Using an electronic feedback technique and hand-held adaptor, this methodology has been applied to determine the volumes of both simple and complex cavities including castings of human nasal passages. The aim of this research is the development of a laboratory system of measurement of nasal passage volume in live humans. This will allow the determination of volume changes resulting from various stimuli and provide further insight into the heat and mass transport phenomena that occur in upper respiratory airways.

U48. New architecture for sub-band coding. Takashi Yazu and Makoto Morito (Research Laboratory, OKI Electric Industry Co., Ltd., 550-5 Higashi-asakawa-cho, Hachioji-shi, Tokyo, 193 Japan)

Most sub-band coders (SBC) have been realized by quadrature-mirror filter (QMF) with frequency shift and low-pass filter (LPF). However, the QMF method has the following problems: (i) Band separation is limited to the power of 2; (ii) arithmetic errors are accumulated because of the cascade connection of QMF; and (iii) it is necessary to adjust the

time delay if the number of QMF connections is different. This paper presents a new architecture for SBC with flexible band assignment, less arithmetic calculations, and small storage memory. In this SBC architecture, a finite impulse response filter (FIRF) is used instead of the QMF. The center frequency of each band is selected according to the equation $F_k/F_s = D \cdot n/m$, where F_s is the sampling frequency of the input signal, F_k is the center frequency of the k th band, D is the decimation factor, and m, n are integers. According to this relationship, frequency shift and the LPF can be performed at the same time. In this case, data memory for the FIRF is drastically reduced because the input of the FIRF is the same for every band. With the new SBC architecture, the 16- to 12-kbps sub-band coder/decoder was implemented on a single DSP. Band selection and suitable bit allocation are discussed.

U49. Using dynamic time warping to formulate duration rules for speech synthesis. Marion J. Macchi, Murray F. Spiegel, and Karen L. Wallace (Bell Communications Research, Morristown, NJ 07960-1910)

Dynamic time warping (DTW), a speech recognition technique, was used to time align long-duration syllables with shorter-duration renditions of those syllables. A male speaker of English produced stressed syllables with long duration, like "libe," and the same syllables with shorter durations in nonsense words like "libous," "libegous," "libegass," and "libotous." Syllables were formant tracked and a frame-by-frame Euclidean bark-formant distance between each long syllable and each of its shorter counterparts was computed. DTW was applied to the resulting distance matrix. The DTW path, referenced to the long syllable, indicates which portions of the long syllable are shortened (and by how much) to match a shorter version of the syllable. The DTW paths showed regions of maximal and minimal shortening. These reactions were approximately aligned across syllable durations. The DTW paths for syllables of different durations were distinguished from one another primarily in regions of maximal shortening. The regions of maximal shortening were aligned with regions in the syllable with little formant movement. Consequently, a spectral-movement time function derived from the templates in an inventory could serve as the basis for duration rules for demisyllable speech synthesis. Further, this technique provides a method for making duration measurements automatically.

U50. Mixed spectral representation—Formants and LPC coefficients. Joseph P. Olive (AT&T Bell Laboratories, Murray Hill, NJ 07974)

A cascade formant model is well suited to describe certain speech segments, such as vowels and vowel-like sounds. The formant model is also useful because the relationship between formants and the vocal tract configurations are well understood; however, this model is not adequate for other speech sounds, such as stops, fricatives, nasals, etc. On the other hand, LPC analysis, of sufficiently high order, can adequately describe the spectrum of any speech sound, but the relationship between the LPC parameters and the spectrum or vocal tract configuration is not obvious. This paper describes a speech analysis/synthesis scheme that uses both formants and LPC parameters for different sections of a speech signal. Thus, in some regions, the benefit of the formant model can be utilized, while, in other regions, the LPC representation can be used to obtain a good description of the speech spectrum. The analysis algorithm resolves the problem of discontinuities that arise from using the two different spectral representations. The method of speech analysis described in this paper produces resynthesized speech of the quality of multipulse LPC.

U51. Speechreading sentences I: Development of a sequence comparator. L. E. Bernstein (Center for Auditory and Speech Sciences, Gallaudet University, Washington, DC 20002), Marilyn E. Demorest (Department of Psychology, University of Maryland—Baltimore

County, Catonsville, MD 21228), and Silvio P. Eberhardt (Jet Propulsion Laboratory, Pasadena, CA 91109)

Previous research suggests that many lexical errors in the speechreading of sentences can be explained in terms of visual phonemic errors. However, description and quantification of perceptual errors at the phonemic level requires specification of stimulus-to-response alignments. Because speechreading produces numerous errors, including phoneme insertions, deletions, and/or substitutions, alignment is a nontrivial problem. This paper describes development of a sequence comparator that can be used to obtain alignments automatically for phonemically transcribed sentences. The comparator employs a weights matrix that reflects presumed visual distances between all possible segmental stimulus-response pairs to find the alignment that minimizes overall stimulus-response distance. Initially, the comparator used weights based on viseme groupings, but these weights resulted in multiple, equal-distance, alternative alignments. More effective weights were obtained empirically via multidimensional scaling of phonemic confusions. Vowel data were obtained from Montgomery and Jackson [J. Acoust. Soc. Am. 73, 2134-2144 (1983)] and consonant data from a nonsense syllable identification task, which employed 22 consonants spoken by the same talkers who produced the sentence stimuli for this study [Bernstein *et al.*, J. Acoust. Soc. Am. 85, 397-405 (1989)]. [Work supported by NIH.]

U52. Speechreading sentences II: Application of a sequence comparator to data on CID sentences. Marilyn E. Demorest (Department of Psychology, University of Maryland—Baltimore County, Catonsville, MD 21228) and L. E. Bernstein (Center for Auditory and Speech Sciences, Gallaudet University, Washington, DC 20002)

When subjects speechread sentences, their performance is typically evaluated in terms of the number of words or keywords correct. A comparator that aligns stimulus and response sequences is being used as a heuristic for studying relationships between the lexical and the phonemic level of speechreading. Toward this end, a corpus of response sequences, previously analyzed in terms of words correct, was reanalyzed with the comparator. Stimuli were 50 CID sentences spoken by a male and a female talker and recorded on video laserdisc. Subjects were normal-hearing college students. One result showed that when sentences are short, computed visual stimulus-response similarity is well-correlated with number of words correct. But with few exceptions, when sentences are long, the correlation is reduced, suggesting that the two measures provide complementary information. The sequence comparator appears to be a useful tool for elucidating patterns of speechreading performance. [Work supported by NIH.]

U53. Investigating randomness in foot timing patterns in English. Briony Williams and Steve Hiller (Centre for Speech Technology Research, 80 South Bridge, Edinburgh EH1 1HN, Scotland)

Isochrony has been considered only in terms of stressed syllables. However, it may also be a random property of unstressed syllables, and a control experiment was deemed necessary. A hand-transcribed database of 98 sentences, each produced by three speakers, formed the input to an algorithm calculating durations of feet, number of syllables per foot, and mean syllable duration within each foot. In each output dataset, feet were based on one of the following: stressed, tense, unreduced, random, or arbitrary syllables (the latter based on ordinal numbers of syllables within the utterance). Calculations were made of the correlations between foot duration and number of syllables per foot, and between foot duration and mean syllable duration. The first correlation was significant for all foot types; the second was significant (and negative) for all except the random and arbitrary types. The conclusion was that, although the mechanism of the tendency toward isochrony had by no means been discovered, it had been shown that the tendency was nonrandom and was due to linguistic rather than arbitrary factors.

Session V. Engineering Acoustics IV: New Transducer Materials

James M. Powers, Chairman

*Naval Underwater Systems Center, New London, Connecticut 06320***Chairman's Introduction—8:25*****Invited Papers*****8:30****V1. Recent developments in piezoelectric composites for transducer applications.** Robert Y. Ting (Underwater Sound Reference Detachment, U. S. Naval Research Laboratory, Orlando, FL 32856-8337)

The design trend for future sonar transducers is to emphasize the utilization of hydrostatic-mode sensing, as opposed to the classical 31- or 33-mode configuration in a Tonpiltz transducer. The conventional PZT-type ceramics cannot meet this need because they have very low hydrostatic piezoelectric coefficients. Piezoelectric PVdF polymers are attractive for this type of applications because of their high sensitivity, but they are limited by their poor temperature stability, low dielectric constant, and planar anisotropy. Piezoelectric composites have therefore been considered as alternate transduction materials that can be used for hydrostatic-mode sensing. Recent developments in ceramic/polymer composites having 1-3 and 0-3 connectivity patterns and nonferroelectric glass/ceramic composites will be discussed. Some test results on the acoustical performance of prototype hydrophones will also be presented to demonstrate the potential of these new materials for underwater acoustical applications. [Work supported by ONR.]

8:55**V2. Ceramic-air composites for hydrostatic sensing.** Manfred Kahn (Naval Research Laboratory, Code 6374, Washington, DC 20375-5000)

The hydrostatic response (d_h) of conventional PZT ceramic is only about 10% to 30% of its biaxial (d_{33}) response. This is expressed in $d_h = d_{33} (1-2 \gamma)$, where γ is Poisson's ratio. Higher d_h values can then be obtained by lowering the Poisson's ratio of the ceramic. This can be accomplished, without lowering the d_{33} value significantly, by introducing into the ceramic arrays of voids, ordered in such a manner that void-free columns of ceramic are maintained between the voids. The deposition of fugitive ink through photolithographically prepared screens onto unfired ceramic tapes, together with stacking, laminating, and subsequent firing, provide a method for the controllable preparation of PZT sensors. These not only exhibit d_h values of ≥ 200 pC/N, but the voids also control the capacitance, so that g_h values of ≥ 0.03 Vm/N are obtained. Capacitance values above 400 pF in a $0.5 \times 0.5 \times 0.1$ -in. sensor chip permit some cable capacitance and the use of bipolar rather than of FET preamplifiers. Hydrostatic tests of such devices have shown these to be also rugged, easy to connect and use, and capable of withstanding significant hydrostatic pressures.

9:20**V3. Monolithic piezoelectric transducers.** W. B. Harrison (Honeywell Ceramics Center, 5121 Winnetaka Avenue North, New Hope, MN 55428)

This paper will review new piezoelectric transducers concepts where layers of lead zirconate titanate (PZT) have been electroded and then cofired into a monolithic structure that was ready for poling and use in various types of transducers. These techniques were initially developed for underwater military applications, but are now ready to be applied to commercial transducers. Several methods for constructing these transducers will be reviewed where either tape processed or more conventionally pressed PZT shapes were used. Typical applications and advantages for this technology in several acoustic transducers will also be described. Relationships between the thermal expansion characteristics of thickness and transverse mode transducers will also be presented along with data on a new, more temperature stable PZT composition. [Work supported, in part, by Office of Naval Research.]

V4. Electrostrictive materials. Robert E. Newnham and Qichang Xu (Materials Research Laboratory, Pennsylvania State University, University Park, PA 16802)

Two types of electrostrictive materials will be described: (1) relaxor ferroelectrics with diffuse phase transitions, and (2) antiferroelectric ceramics with a field-induced ferroelectric phase. Transducers made from relaxor ferroelectrics such as lead magnesium niobate (PMN) and lead zinc niobate (PZN) are operated above T_c in a broad transition range characterized by the presence of nanometer-size microdomains that impart large electromechanical coupling coefficients to the ceramic. These relaxor materials show no piezoelectric effect under ac drive because of the absence of any net spontaneous polarization, but microdomain fluctuations give rise to large electrostrictive strains ($\sim 10^{-3}$) with no hysteresis. Under dc bias, PMN ceramics show piezoelectric d_{33} coefficients as large as 1500 pC/N, three times those of PZT. The second type of electrostrictive transducer is based on a field-induced phase change from an antiferroelectric structure to a ferroelectric phase. Ceramics in the $\text{PbZrO}_3\text{-PbTiO}_3\text{-PbSnO}_3$ ternary system are used as bistable transducers for digital motions where shape-memory hysteresis can be an advantage. Other uses for electrostrictive transducers include adaptive optic systems, micropositioners, and smart transducers in which the size of the piezoelectric coefficient can be adjusted with a dc bias field.

10:10

V5. Rare earth magnetostrictive materials. Arthur E. Clark (White Oak Laboratory, Naval Surface Warfare Center, Silver Spring, MD 20903-5000)

The rare earth elements Tb and Dy possess the largest known magnetostrictions by far (nearly 1%). While these giant magnetostrictions are currently available only at cryogenic temperatures, magnetostrictive strains as high as 0.3% have been obtained at room temperature in compounds of the rare earths. In this presentation, focus will be on the recently developed Terfenol-D compounds prepared by free standing zone (FSZ) and modified Bridgman (MB) techniques. Because near balance of the magnetic anisotropy can be achieved in these compounds, huge magnetostriction "jumps" can occur. During these jumps, energy is transformed between an internal magnetic state to an external mechanical state causing large spontaneous magnetostrictions. Special emphasis in the presentation will be on: (1) magnetostrictive properties at low and high temperatures, (2) dependence of the magnetization and magnetostriction on external stress, (3) preparation techniques and growth morphology of the highly magnetostrictive compounds, (4) Delta E effect, and (5) theoretical models. Recent designs of low-loss transducer modules using high-strain rare earth alloys will be presented. [Work supported by the NSWC Internal Research Program and the Office of Naval Technology.]

10:35

V6. Silicon transducer technology. W. S. Henrion (Triton Technologies, Inc., 1301 Capital of Texas Highway South, Austin, TX 78746)

The use of integrated circuit technology to build mechanical structures by the use of silicon micromachining has gained wide acceptance for some sensor types over the past 10 years. The principal application for silicon micromachining has been for pressure transducers used in the automotive and medical markets. The use of silicon micromachining technology in acoustical sensors has been somewhat limited, although some microphones and accelerometers have been built using this technology. This paper will review the micromachining technologies with specific application to the acoustics and acceleration sensors. Examples will be given for microphones, accelerometers, and pressure transducers with the material process technology used for their manufacture. Projections are made of future trends in silicon micromachining technology for acoustic transducers. The electronic interface to these acoustic and acceleration transducers are explored.

Contributed Papers

11:00

V7. The spherical wave absorption coefficient of a patch of material. Matthew A. Nobile (IBM Acoustics Laboratory, Department C18, Building 704, P.O. Box 950, Poughkeepsie, NY 12602)

Recently, the concept of a *spherical wave absorption coefficient*, α_{spher} , was introduced [M. A. Nobile, Proc. NOISE-CON 88, 567-572 (1988)] to account for the fact that when spherical waves are incident upon an acoustical material, the absorption coefficient is not the same as it would be under (less realistic) plane-wave incidence. In fact, α_{spher} is not an

intrinsic property of the material but depends on the distance of the sound source from the material. In this paper, an expression for the spherical wave absorption coefficient of a finite-size patch is derived (in the cylindrical coordinate system considered, the patch is either a circular or annular disk). As will be shown, this quantity can be approximated experimentally from measurements of the reflection coefficient at several oblique angles over the surface of the sample using the two-microphone transfer function method in a hemianechoic environment [an extension of M. A. Nobile, Proc. NOISE-CON 87, 611-616 (1987)]. This can be the basis for a simplified method of comparing the absorptive properties of two materials.

11:15

V8. Smart acoustically active surfaces. F. Douglas Shields, James E. Hendrix, and L. Dwyann Lafleur (National Center for Physical Acoustics, University, MS 38677)

Bilayered transducing surfaces have been constructed and used to control the reflection of sound in both liquids and gases. The acoustic signal generated in one of the layers is amplified, passed through a digital filter, and used to drive the second layer with the proper phase and amplitude so as to eliminate the reflections of single-frequency tone bursts. The tone burst frequencies were between 20 and 40 kHz in liquid and between 5 and 20 kHz in air. Feedback is controlled by the digital filter so as to eliminate instabilities. A theoretical analysis predicts the performance of such smart surfaces, but as yet there are discrepancies between theory and experiment. [Work supported by ONR.]

11:30

V9. Ultrasonic investigation on different structure and composition copolymers. Jiankai Hu, Zhou Jing, and Xu Wenying (Department of Electronics, University of Science and Technology of China, Hefei, Anhui, People's Republic of China)

Ultrasonic attenuation and velocities have been measured for three kinds of copolymers: MMA-BMA, MMA-MAA, and ST-MAA containing different kinds of cross-linkers. Experimental results indicate that ultrasonic attenuation increases with increasing BMA content in MMA-BMA, with adding cross-linkers EGD or BA in MMA and in ST-MAA;

velocity decreases with increasing BMA content in MMA-BMA. The temperature dependences of ultrasonic attenuation in MMA-BMA show that the glass transition temperature T_g was decreased with increasing BMA content, and T_g could be easily determined with ultrasonic method. The results have been explained by thermoelastic mechanism and scattering mechanism.

11:45

V10. Ultrasonic investigation of a BiSrCaCuO high T_c superconductor. Jiankai Hu, Qianlin Zhang, Deng Tingzhang, and Weili Cai (Department of Electronics, University of Science and Technology of China, Hefei, Anhui, People's Republic of China)

The temperature dependences of ultrasonic attenuation and velocity in a high T_c superconductor BiSrCaCuO were measured by the ultrasonic pulse echo method, in the temperature range 77–300 K. Ultrasonic attenuation anomalies were found at 95, 135, and 250 K; the slope of ultrasonic velocity was changed at these temperatures respondingly. There are two superconductive phases in BiSrCaCuO samples at 87 and 120 K found from x-ray examining. The ultrasonic anomalies at 95 and 135 K are related to the superconductive transition at 87 and 120 K resulting from microstructure instability. The same phenomenon has also been found by internal friction experiments [H. Shen *et al.*, Proc. of 2nd Internal Friction and Ultrasonic Attenuation Conference of China, C63 (1988)]. An ultrasonic anomaly at 250 K is a relaxation process, similar to the mechanism in YBCuO [M-F. Xu *et al.*, Phys. Rev. B 37, 3675 (1988)].

WEDNESDAY MORNING, 24 MAY 1989

NEWHOUSE II, ROOM 254, 8:30 A.M. TO 12:30 P.M.

Session W. Bioresponse to Vibration II, Psychological Acoustics IV, and Physiological Acoustics IV: Vibrotaction—Physiological, Psychophysical, and Applied Aspects

Ronald T. Verrillo, Chairman

Institute for Sensory Research, Syracuse University, Syracuse, New York 13244

Chairman's Introduction—8:30

Invited Papers

8:35

W1. Four channels mediate vibrotaction: Facts, models, and implications. S. J. Bolanowski, Jr. (Institute for Sensory Research, Syracuse University, Syracuse, NY 13244)

Four separate pathways or channels apparently mediate tactile (i.e., mechanical) stimuli. Each channel, as defined psychophysically by its frequency characteristics in response to vibratory stimuli and physiologically by additional considerations such as receptive field sizes and adaptation rates to ramp-and-hold stimuli, operates over a wide range of frequencies and intensities. The psychophysical channels are: Pacinian or P; non-Pacinian I or NPI; non-Pacinian II or NP II; and non-Pacinian III or NP III. As defined physiologically and anatomically, the channels are hypothesized to be, respectively: Pacinian (Pacinian corpuscles) or P; rapidly adapting I (Meissner corpuscles) or RA I; slowly adapting II (Ruffini endings) or SA II; and slowly adapting I (Merkel-cell neurite complexes) or SA I. Because the channels are overlapping in their operating ranges, it is proposed that under certain conditions (e.g., low stimulus intensities) a single channel may mediate taction, while other conditions require that more than one channel contributes to tactile sensations and to overall perceptual impressions. From the standpoint of applied aspects such as the development of tactile aids for the deaf, it should be of interest to determine if the channels can be used independently and to determine the types of interactions that may occur for the purpose of enhancing information transfer via this sensory system.

W2. Vibrotactile measurement procedure for assessing peripheral neuropathies. A. J. Brammer and J. E. Piercy (Division of Physics, National Research Council, Ottawa, Ontario K1A 0R6, Canada)

It has been suggested that some peripheral neuropathies may be detected earlier by vibrotaction than by more traditional methods (viz., nerve conduction). Most commercial apparatus for this purpose is designed primarily for use on any flesh surface, and only records the sensitivity of one mechanoreceptor type. Recent laboratory studies have identified a technique for establishing, independently, the sensitivity of each of the three receptor populations essential for the sense of touch at the fingertip, with minimal effect from confounding variables. A portable measurement system using this technique has now been developed for clinical use or field surveys, which enables the thresholds of untrained subjects to be determined rapidly, by employing audiometriclike procedures. The data are conveniently summarized by "tactograms," or "tactilograms," in which the change in vibrotactile threshold from the mean value recorded in normal hands, at each frequency, is expressed graphically, in a manner similar to an audiogram. Results from persons with diagnosed or suspected neuropathies have revealed three patterns of threshold shift, at least one of which is receptor specific.

9:35

W3. Clinical aspects of vibrotactile devices with deaf children. Mary Joe Osberger and Amy M. Robbins (Department of Otolaryngology, Indiana University School of Medicine, Indianapolis, IN 46223)

Longitudinal data will be presented on the speech perception and production abilities of 15 deaf children who use a vibrotactile aid (Tactaid II). Their performance is being compared to that of deaf children who use either a single- or multichannel implant. Initial findings indicate that the performance of the subjects with implants, even those with single-channel devices, is superior to that of the subjects who use the tactile aid. The subjects who achieved the highest scores with the tactile aid tended to have more sophisticated cognitive and linguistic skills that permitted them to develop strategies to decode the vibratory patterns. The importance of prior knowledge of the language in decoding information perceived via the skin is also supported by the data collected from postlingually deaf adults using the same tactile aid. The implications of generalizing data collected on adults to prelingually deaf children will be discussed. [Work supported by NIH.]

10:05-10:15
Break

Contributed Papers

10:15

W4. Independence of spatial and temporal tuning in the NP I cutaneous mechanoreceptor system. Clayton L. Van Doren (Department of Biomedical Engineering, Case Western Reserve University, Cleveland, OH 44106)

Results from a previous study that measured tactile detection thresholds for spatiotemporal sinusoids suggest that the spatial and temporal tuning of the P and NP I cutaneous mechanoreceptor systems are independent. In other words, provided that the spatial configuration of a vibratory stimulus is independent of temporal frequency, changing the spatial configuration should not change the *shape* of the temporal tuning curve of either system. To test this hypothesis for the NP I system, vibrotactile thresholds were measured as a function of temporal frequency (10, 12, 15, 20, 25, 30, 40, 55, 80, 100, 175, and 250 Hz) for 200-ms stimuli delivered to the thenar eminence via a 0.75-cm² circular contactor, with and without a rigid surround. At frequencies where the threshold was mediated exclusively by the NP I system, the presence of a rigid surround did not significantly alter the shape of the temporal tuning curve, but did change the overall sensitivity. That is, NP I thresholds were uniformly higher without the surround than with the surround. Similar results were obtained when the experiment was repeated on the forearm. The observed invariance in the shape of the low-frequency tuning curve is consistent with the spatiotemporal independence hypothesis, but is consistent with conclusions from prior studies. The apparent discrepancy will be discussed.

10:30

W5. Identifying possible neural codes for vibrotaction using the theory of temporal summation. C. M. Checkosky and S. J. Bolanowski, Jr. (Department Surgery, University of Rochester Medical School, Rochester, NY 14642 and Institute for Sensory Research, Syracuse University, Syracuse, NY 13244)

The theory of temporal summation refers to the psychophysical phenomenon in which threshold decreases with increasing stimulus duration. The effect is found in audition and is central. Vibrotactile thresholds mediated by the Pacinian-capsule (PC) population (the "P" channel) also exhibit temporal summation. Assuming that the tactile summator is also central, it may be possible to determine the neural code for threshold used by the P channel. Theoretically, temporal summation predicts a decrease in threshold at a rate of -3 dB/doubling of duration from 10 to 100 ms. Thus by integrating the response from single fibers for various response criteria, those not having the potential for temporal summation can be rejected as possible coding schemes. PCs isolated from cat mesentery were stimulated with bursts of vibrations (300 Hz) having durations from 10 to 1000 ms, the intensity of the stimuli adjusted for fixed criterion responses of 1, 2, and 4 impulses/burst and 1 impulse/cycle. Data were sampled over the entire burst duration, in effect analyzing the response with an ideal summator. The decreases in intensity for increases in duration from 10 to 100 ms were -0.9 , -1.2 , -2.3 , and -0.2 dB/doubling, respectively, with neither the 1 impulse/burst nor the impulse/cycle criteria showing significant summation. The other two criteria show modest sum-

mation, indicating that they have the potential for being the neural code for threshold in the P channel. [Work supported by NS23933.]

10:45

W6. Vibrotactile forward masking in young and old subjects. G. A. Gescheider and A. Valetutti, Jr. (Hamilton College, Clinton, NY 13323)

Thresholds for the detection of a 50-ms test stimulus were measured as a function of the time interval between the offset of a 500-ms masking stimulus and the onset of the test stimulus (Δt). The frequency of the masker and the test stimulus was the same during a particular testing session and was either 20 or 250 Hz. The stimuli were presented through a 2.9-cm² circular contactor applied to the thenar eminence. Thresholds were measured by a two interval-forced choice tracking procedure. A group of seven young subjects (average age 21 years) and a group of older subjects (average age of 67 years) were tested. The amount of forward masking expressed as threshold shift decreased as Δt increased. At all values of Δt , the older subjects exhibited significantly more masking than did the young subjects. The forward masking functions and the effects of age were essentially the same for stimuli that primarily affect the Pacinian system (250 Hz) and those that primarily affect non-Pacinian systems (20 Hz).

11:00

W7. Vibrotactile threshold for subjects with peripheral neuropathies. L. H. Novick, R. T. Verrillo, S. J. Bolanowski, Jr., A. Ecker, and G. A. Gescheider (Institute for Sensory Research, Syracuse University, Syracuse, NY 13244)

Thresholds for detecting vibrotactile stimuli applied to the thenar eminence through a 2.9-cm² contactor were measured for patients having reflex sympathetic dystrophy (RSD, $n = 7$) and carpal tunnel syndrome

(CTS, $n = 2$). Thresholds were measured with a two-interval, forced-choice tracking procedure. The vibratory stimuli consisted of 700-ms sinusoidal bursts with rise-fall times of 37 ms. The frequency of the test stimuli ranged from 3–300 Hz. Three of the RSD patients demonstrated a 20- to 30-dB loss of sensitivity at all frequencies for their affected hands. These patients had normal sensitivity in their unimpaired hands. The other four RSD patients demonstrated no loss in sensitivity in neither their affected nor normal hands. The two patients tested with CTS also had normal threshold functions for both their affected and unaffected hands. The results suggest that the clinical diagnosis of RSD subsumes several pathological conditions one of which may or may not be a decrease in threshold sensitivity. The fact that CTS patients may not experience impaired threshold suggests that this neuropathy may not affect nerve fibers mediating low-threshold tactile sensitivity.

11:15

W8. A comparison between the growths of loudness of vibrotactile pulse and sinusoidal stimuli. Amy A. Collins and Roger W. Cholewiak (Department of Psychology—Green Hall, Princeton University, Princeton, NJ 08544-1010)

The growths of loudness of pulsatile and sinusoidal stimuli were examined by the method of absolute magnitude estimation. Fourteen observers judged 32-pps pulse and 250-Hz sine stimuli on two sites on the palm: the thenar and hypothenar eminences. Estimates revealed that pulse stimuli were perceived as being louder than the sine stimuli on both sites. Power functions fit to the sine data had exponents of 0.84, while those fit to the upper limbs of the pulse data had slopes of 0.82. Cross-quality matches between the two types of stimuli were made by nine of these observers. The group matches were not found to be internally consistent. That is, those made when the sine was the standard were not equivalent to matches made when the pulse was the standard. Such consistency was apparent, however, in some individual observer's data. In addition, slopes of matching functions were often well predicted by magnitude estimates using a procedure based upon the absolute scaling hypothesis. Intercepts were also predicted in some individual cases. [Work supported by NANCDS.]

11:30–12:30

Discussion Group

Session X. Musical Acoustics II: Piano Acoustics

William Y. Strong, Chairman

*Steinway and Sons, Steinway Place, Long Island City, New York 11105****Invited Papers*****8:30****X1. Piano history—Science and art.** William Y. Strong (Steinway and Sons, Steinway Place, Long Island City, NY 11105)

The piano enjoys a unique position in both the concert hall and the home. It has been in existence for well over 200 years and was the subject of much development activity during the middle 1800s. Competition was fierce at the time and innovation was crucial to a company's survival. Artists placed musical demands on the instruments, designers countered with their own ideas and capabilities, and reputations were made on the resulting instrument. Some of the great minds of the day focused on the piano acoustics challenge, perhaps most notably Hermann Helmholtz, a name familiar to all acousticians. This talk will present a few interesting historical details of the liaison between science and art in the design of the piano.

8:50**X2. Vibration characteristics of rectangular test soundboards.** Hideo Suzuki (Ono Sokki Company, Limited, 2-4-1 Nishishinjuku, Shinjuku-ku, Tokyo, 163 Japan)

Since a piano soundboard has a very complex structure, it is difficult to separately study the effect of each parameter (such as the board thickness, rib dimensions, bridge dimensions) on the vibration characteristics of the whole soundboard. To more effectively study the effect of these parameters, four test soundboards were built. A much simpler structure was used to study the vibration characteristics of a spruce panel reinforced with several spruce ribs. Driving point inertance functions at different points were compared among the four soundboards. Mode numbers and modal patterns were also compared. It is interesting that the vibration at a point on (or near) a rib is quite different from that on a point away from ribs in the frequency range above the coincidence frequency. [This study was performed while the author was at CBS Technology Center, Stamford, CT and supported partially by Steinway and Sons, Long Island City, NY.]

9:20**X3. Wood for piano soundboards.** Daniel W. Haines (School of Engineering, Manhattan College, Riverdale, NY 10471)

The wood soundboard of the piano is among the most important components of the instrument. It is essential for projecting the desired volume of sound, and the soundboard material strongly influences the quality of the sound. As it is for the sound radiating elements of many string instruments, spruce is the unanimous choice of wood for piano soundboards. The reasons for the use of spruce and selection criteria for superior performance will be discussed.

9:50**X4. The mathematics of piano tuning.** Albert E. Sanderson (Inventronics, 171 Lincoln Street, Lowell, MA 01852)

It is a well-known fact that piano strings vibrate inharmonically—the partials actually run sharp by amounts that vary from piano to piano and from note to note on any one piano. This affects the tuning significantly, and it is correct to say that no two pianos should be tuned exactly alike. This paper presents a mathematical solution to the problem of tuning such instruments for typical classes of inharmonicity versus note-number curves, the most important case being a straight-line logarithmic increase of about 3:1 per octave. Equal temperament is literally impossible in this case, and if one forces it with an electronic tuning, the instrument sounds very unmusical. The octave is no longer exactly 2:1, and must be redefined to allow for different types of octave partial matching. More general definitions of equal temperament and octaves lead to a tuning solution in the form of a difference equation. Although no two octaves or even semitones are exactly the same width, they are consistently wide by an amount that grows wider as the inharmonicity rises. Comparison with pianos tuned aurally shows that fine piano tuners have been solving this equation subconsciously for centuries.

X5. Nonlinear modeling of piano hammer-string interaction. Donald E. Hall (Physics Department, California State University, 6000 J Street, Sacramento, CA 95819)

Measurements of compliance on real piano hammers show that they are not well represented by Hooke's law, and it has been found that theoretical models with linear compliance have only modest success in accounting for measured string vibration spectra. Two new computer programs now perform numerical integrations of the interaction process for hammers with nonlinear compliance. One treats the string as a pair of digital delay lines, and is limited to perfectly flexible strings. The other represents the string by its normal modes, and can include string stiffness. Results from these two programs agree well with each other, and provide improved agreement with laboratory measurements.

Contributed Papers

10:50

X6. Timing in skilled piano performance. Caroline Palmer (Psychology Department, Ohio State University, Columbus, OH 43210)

Skilled music performance is characterized by expressive variations that communicate the performer's interpretations of structural content to a listener. Four experiments are described that address the procedures that pianists use to govern the assignment of structural interpretation to timing variations. Six piano performances of the same music were recorded on a computer-monitored acoustic piano. Three timing methods were measured in each performance; chord asynchronies, rubato patterns, and overlaps (legato/staccato patterns). Each pattern corresponded to the pianists' notated interpretations of musical structure. Experiment 2 investigated the development of the timing methods during learning. Each pianist's successive performances of an unfamiliar musical excerpt demonstrated the methods: Students' performances showed an increase in each method, while experts' performances stayed constant (with more of each method). In experiment 3, each pianist performed different interpretations of the same music. The same methods (applied in different amounts) were used in each performance, and increased in exaggerated performances of the same structural interpretation. These results support a mapping of musical intentions to performance timing through a set of procedures shared by many pianists. [Supported by NSF, MIT Media Laboratory and Psychology Department, Cornell University.]

11:05

X7. Further tests of "composers' pulses" in computer performances of piano music from the classical period. Bruno H. Repp (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695)

Composers' pulses are patterns of expressive microstructure (i.e., timing and amplitude modulations) said to convey individual composers' personalities when implemented in computer performances of their music [see M. Clynes, in *Studies of Music Performance*, edited by J. Sundberg (1983), pp. 76-181]. The present research continued the perceptual evaluation of such pulses, which so far has yielded inconclusive results [B. H. Repp, *Music Perception* (in press)]. The initial bars of five piano pieces by each of four composers (Beethoven, Haydn, Mozart, and Schubert) were played on a computer-controlled electronic piano with each of the four composers' pulses (following Clynes' specifications as closely as possible) and also in a deadpan version. Listeners judged to what extent each computer performance conveyed the composer's characteristic expression, relative to the deadpan version. Pilot results provide occasional positive support for the Beethoven and Haydn pulses, but none at all for the Mozart and Schubert pulses. Additional data will be reported from musically trained subjects who also listened to computer performances in which the timing and amplitude components of the pulses were applied separately. [Work supported by NIH-BRSG.]

11:20

X8. Properties of piano hammers. Anders Askenfelt (Department of Speech Communication and Musical Acoustics, Royal Institute of Technology, Stockholm, Sweden)

Measurements of the compliance and loss factors for raw hammer felt and completed hammers have been recorded. The results of these measurements will be presented and their implications discussed. The potential for new materials for use in piano hammers is suggested and possible candidates for these new materials will be discussed.

Session Y. Physiological Acoustics V: From the Ear Canal to the Cortex

Robert V. Shannon, Chairman

Boys Town National Institute, 555 North 30 Street, Omaha, Nebraska 68131

Contributed Papers

8:30

Y1. On ear canal measurement techniques applied to hearing development. Douglas H. Keefe and Robert Ling (Systematic Musicology Program, School of Music, DN-10, University of Washington, Seattle, WA 98195)

Few data exist on the functional development of the human external and middle ear, yet constraints imposed by these structures limit the sound power transmitted to the cochlea. One measure of hearing development is the middle ear reflectance, the fraction of incident sound power reflected at the eardrum. Single and double microphone techniques measure the impedance at the tip of an assembly snugly inserted into the ear canal. This impedance is of limited usefulness, since it depends not only upon the middle ear reflectance, but also upon insertion depth, variations in cross-sectional ear canal area, and ear canal curvature. A method has been constructed to assess the joint influence of the ear canal area function and curvature with a spatial resolution along the ear canal of 4 mm ($cT/2$, for phase velocity c and sample period T). The reflectance at the assembly tip is transformed to that at an ear canal location just outside the eardrum. The method is also relevant for constructing the output impedance of the ear, as seen by a source on the eardrum. This technique makes no use of invasive probe microphones, has a frequency resolution of 40 Hz, and is well suited for use with infants. Preliminary data on tubes of known geometry will be discussed. [Work supported by NIH.]

8:45

Y2. Revision of a two-piston eardrum model to account for real ear canal geometries. R. Michael Stinson (Division of Physics, National Research Council, Ottawa, Ontario K1A 0R6, Canada)

The acoustical behavior of the middle ear can be conveniently summarized using acoustical network representations, analogous to electrical circuits. Previous middle ear networks, though, have been derived assuming that the ear canal could be treated as uniform in cross section and terminated perpendicularly by the eardrum. At frequencies above about 5 kHz, it is necessary to account for the actual ear canal geometry (varying cross-sectional area along canal length). In this paper, the two-piston representation of Shaw [J. Acoust. Soc. Am. Suppl. 1 62, S12 (1977)] is incorporated into a horn equation description of the sound field in ear canals. Several of Shaw's network parameters must be revised so that eardrum impedances and ear canal standing wave ratios will be correctly predicted when this network is used in conjunction with realistic canal geometries. The analysis makes use of recent measurements of human ear canal geometry.

9:00

Y3. Directional dependence of free-field-to-eardrum transformations in cats. J. Jeremy Rice, Brad May, George A. Spirou, and Eric D. Young (Center for Hearing Sciences and Department of Biomedical Engineering, Johns Hopkins University, Baltimore, MD 21205)

The directional dependence of the transfer function from free-field plane waves to a point near the tympanic membrane was measured in anesthetized domestic cats. A probe tube microphone was placed from a ventral approach, which minimally affects the head and pinnae. Acoustic measurements were taken using a quasianechoic method in which an impulsive stimulus was presented and room reflections were rejected using a Kaiser window. The transfer functions are similar to those reported earlier [A. D. Musicant *et al.*, 12th ICA, Toronto (July 1986)]. They can be broken into three distinct frequency ranges: A low-frequency range (< 5 kHz), where interaural level differences vary smoothly with azimuth; a midfrequency range (5–18 kHz), where a prominent spectral notch is observed; and a high-frequency range (> 18 kHz), where the spectrum varies greatly with source location. The frequency of the midfrequency notch is a robust cue for sound source direction; it varies with both azimuth and elevation in a monotonic fashion in the frontal field. Source direction is uniquely determined by combining the midfrequency notch cue from both ears. These results suggest that simple consideration of interaural level differences may not be sufficient for the study of sound localization at high frequencies. [Work supported by Grant No. NS12524 from NINCDS/NIDCD.]

9:15

Y4. Ear canal cross-sectional acoustic modes. M. T. Friedrich and R. D. Rabbitt (Department of Mechanical Engineering, Washington University, St. Louis, MO 63130)

The cross-sectional acoustic modes for real ear canal geometries are determined using a WKB method in combination with a Rayleigh-Ritz procedure. This model, which applies the usual assumptions of linear acoustics, can be used to analyze the three-dimensional sound-pressure distribution in any long, slender, rigid-walled waveguide. The method is particularly well suited to obtain the higher modes, which are trapped near both ends of the ear canal. The analysis was confirmed by comparing computed results with the exact solutions for fundamental waveguides. Results were also obtained for geometries measured from real ear canals. [Work supported, in part, by the Whitaker Foundation.]

9:30

Y5. An approach to finite-element modeling of the middle ear. Susan M. Funnell and W. Robert J. Funnell (BioMedical Engineering Unit, McGill University, 3775 University Street, Montréal, Québec H3A 2B4, Canada)

It is becoming increasingly clear that the mechanical behavior of the middle ear is more complex than first thought. In particular, recent experimental evidence indicates that the ossicular axis of rotation is not fixed. Finite-element analysis is a method well suited to the detailed analysis of complicated mechanical systems. A finite-element model of the eardrum has previously been developed. This paper presents work on a system developed to model the middle ear ossicles and important soft tissues. Anatomical information is input to computer by digitizing outlines of structures of interest from serial section slides. The resulting three-dimen-

sional volume is discretized into tetrahedral elements using an automatic mesh generator developed to mesh irregularly shaped objects. Preliminary results of analysis on middle-ear structures will be presented and modeling of the middle ear as a whole will be discussed. [Work supported by MRC Canada.]

9:45

Y6. Stimulus frequency otoacoustic emissions: Evidence for multiple sources in the cochlea. Beth A. Prieve and Paul J. Abbas (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

Stimulus frequency otoacoustic emissions (SFE) were measured in humans by recording the total sound-pressure level in the ear canal upon stimulation with continuous pure-tone stimuli across a wide frequency range. SFE are typically not equal in magnitude as a function of frequency. There are frequencies at which large magnitude emissions exist bounded by frequencies at which smaller magnitude emissions exist. SFE can be masked by simultaneous presentation of a second pure tone and the masking functions demonstrate strong frequency dependence. These tuning properties were measured with probes at frequencies evoking large magnitude and those evoking the surrounding smaller magnitude emissions. For each tuning curve, the trip is higher in frequency than that of the probe. As probe frequency is increased, the tip frequency also increases, although not always proportionally. Assuming that the tip of the tuning curve is

indicative of a place on the basilar membrane, the data are consistent with a model in which otoacoustic emissions are generated at more than one place in the cochlea.

10:00

Y7. Effect of contralateral acoustic stimulation on $2f_1-f_2$ acoustic distortion in human ears. Kenneth R. Tough and H. Kunov (Institute of Biomedical Engineering, University of Toronto, Toronto, Ontario M5S 1A4, Canada)

The normal human cochlea is known to exhibit acoustic nonlinearity, observed most easily as a $2f_1-f_2$ distortion product tone in response to two tonal stimuli. This distortion product is physiologically vulnerable, and can be influenced in animals by electrical stimulation of the cochlear efferent nerves. The function of the efferent cochlear innervation is poorly understood but currently revived theory suggests a feedback system may exist between the opposite cochlea, influencing cochlear micromechanics through the outer hair cells [D. O. Kim, *Hear. Res.* 22, 105-114 (1986)]. The influence of a contralateral acoustic stimulus on cochlear nonlinearity was studied in human subjects through the changes observed in the $2f_1-f_2$ distortion product. The influence of interaural acoustic transmission of the stimulus was controlled by using constant SPL stimuli on subjects with varying degrees of sensorineural hearing loss in the stimulated (contralateral) ear. Results indicate a slight suppression in distortion product with no significant frequency tuning. Strong correlation with middle ear reflex thresholds and lack of correlation with sensation level indicate the suppression is not caused by a cochlea-to-cochlea neural mechanism, and is likely caused by the middle ear reflex.

10:15-10:30
Break

10:30

Y8. Characteristics of the $2f_1-f_2$ difference tone in human subjects and in a hardware cochlear nonlinear preprocessing model with active feedback. Frances P. Harris and Eberhard Zwicker (Institute of Electroacoustics, Technical University of München, Arcisstrasse 21, 8000 München 2, Federal Republic of Germany)

Level and phase of the $2f_1-f_2$ combination tone were measured in five human subjects using two methods: Psychoacoustic cancellation and direct cancellation of the distortion-product emission (DPE) measured in the ear canal. A probe assembly, composed of two earphones and a microphone, was placed in the subject's ear canal for the delivery of stimulus tones, f_1 and f_2 , and the $2f_1-2f_2$ cancellation tone and for measurement of DPEs. Frequency of f_1 was 1620 Hz with f_2 either 1800, 1851, 1944, or 2052 Hz. Level of f_1 or f_2 was varied in 5-dB steps from 70-30 dB SPL for each frequency pair. Data from these subjects were compared with similar data measured using the hardware model. DPE cancellation levels were 40-50 dB below comparable psychoacoustical cancellation levels for both data sets, however absolute differences varied by frequency separation and level of the primaries and by method. Dependencies of the level and phase of $2f_1-f_2$ on the frequency separation and level of the primaries were similar for all three methods.

10:45

Y9. Detection of modulation in pulse trains by patients and cochlear implants. Robert V. Shannon (Boys Town National Institute, 555 North 30th Street, Omaha, NE 68131)

Detection of amplitude and pulse-width modulation was measured in patients with the Symbion and Nucleus cochlear implants as a function of

modulation frequency. The resulting temporal modulation transfer functions (TMTFs) are similar to those of normally hearing listeners [N. F. Viemeister, *J. Acoust. Soc. Am.* 66, 1364-1380 (1979)]. Modulation detection functions are basically low pass, with best sensitivity to modulation frequencies of 20-100 Hz and decreasing sensitivity for modulation frequencies above 100 Hz. Modulation detection performance is similar whether the carrier is an analog sinusoidal stimulus or a biphasic pulse train. These results suggest that envelope detection mechanisms are intact in these implanted patients, even though their cochleae are not functioning and the pattern of activation in their auditory nerves are quite different from any normal acoustic pattern of activity. [Work supported by NIH NS24754.]

11:00

Y10. A negative correlation between endocochlear potential and endolymph potassium concentration. Alec N. Salt, Ruediger Thalmann, and John DeMott (Department of Otolaryngology, Washington University, St. Louis, MO 63110)

It is widely accepted that the endocochlear potential (EP) arises from a secretion of K^+ into endolymph by stria vascularis, although, as yet, there is no consensus concerning that cellular basis of this process. Studies in which changes in EP and endolymph potassium (K_e) were recorded during anoxia or treatment with transport inhibitors (e.g., ouabain) have shown that a decline of K_e is associated with the suppression of EP. On the basis of such experiments, it would be presumed that K_e would tend to be positively correlated with the magnitude of EP. On the contrary, a number of procedures have been found in which a decrease of K_e is associated with an increase of EP. Perilymphatic perfusion of solution containing 20 mM K^+ caused a transient increase of EP (by up to 10 mV)

followed by a decline. During this period, K_e recorded by ion-selective microelectrodes typically showed a decrease, followed by an increase. Similarly, iontophoretic injection of Ca^{++} into endolymph produced an EP decrease that was mirrored by a K_e increase. These findings cannot be explained by changes in K^+ transport and/or K^+ permeability alone, but are more likely accounted for by changes in anion permeability. [Work supported by NIH Program Grant No. P01 NS24372.]

11:15

Y11. Voids in the gerbil cochlear nucleus: Is there an anatomical substrate for intensity coding? K. D. Statler, S. C. Chamberlain, R. L. Smith, and N. B. Slepecky (Institute for Sensory Research and Department of Bioengineering, Syracuse University, Syracuse, NY 13244)

The presence of voids in the gerbil cochlear nucleus has been reported by Morest *et al.* [IUPS Satellite Symposium on Hearing Abstract, p. 80 (1986)] and related to some aspects of auditory experience by McGinn and Faddis [Hear. Res. 31, 235-244]. The number, volume density, and void size were investigated in animals of various ages from 18-485 days. These variables were found to increase as a function of age with a maximum volume density of about 1% [Statler *et al.*, ARO Abstracts, p. 209 (1988)]. The spatial distribution of the number of voids in the cochlear nucleus was computed as a function of the incoming neural activity produced by ambient noise over the life of the animal. In a three-dimensional reconstruction, the voids form a contiguous area in the granular layer and posteroventral cochlear nucleus. The anteroventral cochlear nucleus and dorsal cochlear nucleus are spared. The number of voids was found to be a monotonically increasing, negatively accelerated function of the total neural activity. The difference between the void distribution and tonotopic organization of the cochlear nucleus leads to the interesting possibility that the spatial distribution of voids corresponds to the low-intensity portion of the tonotopic regions of the cochlear nucleus and the void distribution reflects an anatomical substrate for intensity coding.

11:30

Y12. Global brain asymmetries in regional cerebral blood flow (rCBF) observed during resting conditions with positron emission tomography (PET): Establishing a baseline for experiments on brain asymmetries and complex sounds in the CNS Project. Judith L. Lauter and Elena Plante (Speech and Hearing Sciences, University of Arizona, Tucson, AZ 85721)

During 1981-1985, the first author and colleagues at the Mallinckrodt Institute of Radiology at Washington University in St. Louis collected a series of 29 PET-scan studies of the brains of 16 normal young adult subjects using the PETT VI and intravenous oxygen-15 to test each subject in multiple 40-s scans, comparing rCBF topography during session-initial and session-final control scans, and under a variety of auditory stimulation conditions. Global hemispheric rCBF in all control scans are currently being measured in this library. Results will be presented for work to date, detailing between- and within-subject comparisons, including patterns to within- and between-session replications. These measurements will provide a baseline for the Coordinated Noninvasive Studies (CNS) Project, in which subjects first examined behaviorally for processing asymmetries (e.g., ear advantages), are then tested with PET to determine the degree of correlation between behavioral and physiological asymmetries. [Work supported by AFOSR.]

WEDNESDAY MORNING, 24 MAY 1989

REGENCY C, 8:40 A.M. TO 12:00 NOON

Session Z. Underwater Acoustics IV: Propagation

Kenneth E. Gilbert, Chairman

National Center for Physical Acoustics, University of Mississippi, University, Mississippi 38677

Chairman's Introduction—8:40

Contributed Papers

8:45

Z1. Efficient computation of ideal starting fields for parabolic equation models. E. Richard Robinson and David H. Wood (Code 3122, New London Laboratory, Naval Underwater Systems Center, New London, CT 06320)

When the Helmholtz equation is replaced by the related parabolic equation, the resulting gain is the relative numerical ease of solving the parabolic equation using a marching type algorithm. However, there is still the problem of providing the initial data—or the starting field for the PE model. Although it is well known that the normal mode starter is the correct initial data, it is not widely used because of the expense involved in the computation. In the present approach, however, the idea is exploited that it is not necessary to compute the normal modes individually to obtain the correct starting field. If the function represented by a weighted sum of the normal modes is considered, then it is found that the correct

starting field can be obtained by computing the $-1/4$ power of the matrix generated from the FFT of the index of refraction. This result is derived and some example starting fields are presented compared to starters computed or approximated in other ways. Further, the effects on PE model outputs using the various starters are demonstrated.

9:00

Z2. A modified k_0 -independent parabolic equation incorporating range refraction. (I. W. Schurman), W. L. Siegmann, and M. J. Jacobson (Rensselaer Polytechnic Institute, Troy, NY 12180-3590)

Parabolic equations currently in use to predict acoustic intensity require a judicious choice of the reference wavenumber k_0 . A new parabolic

equation (PE) is presented here which is independent of k_0 and capable of handling large variations in the index of refraction. This equation is similar to, and ostensibly simpler than, an earlier range refraction PE (RAR-EPE) reported recently by Tappert and Lee. The modified refraction parabolic equation (MOREPE) is obtained by a transformation approach, and operator and multiscale formalisms are also described to validate the equation. Principal properties of MOREPE are discussed. An exact solution with only range variation in sound speed is presented to illustrate differences between standard PE and MOREPE. Propagation examples in range-independent environments demonstrate close agreement between MOREPE and standard PE, while examples with strong range dependence exhibit significant differences between the two equations in their predictions of acoustic intensity. [Work supported by ONR.]

9:15

Z3. Applications of higher-order parabolic equations. Michael D. Collins (Naval Ocean Research and Development Activity, Stennis Space Center, MS 39529)

The parabolic equation (PE) model is very useful for many range-dependent acoustic calculations. However, the PE solution breaks down for propagation at large angles, out to long ranges, and in domains in which sound-speed variations are relatively large. This problem was reduced with the introduction of the wide-angle PE of Claerbout, which is based on a rational linear Padé approximation. Generalizing this approach to a sum of rational linear terms [Bamberger *et al.*, SIAM J. Appl. Math. **48**, 129-154 (1988)] leads to a higher-order PE that is easy to solve numerically. Calculations will be presented to demonstrate that this higher-order PE accurately handles problems involving very wide angles of propagation and large differences in sound speed including elastic wave propagation involving a superposition of both shear and compressional waves.

9:30

Z4. Spectral decomposition of parabolic equation fields. Kenneth E. Gilbert, Xiao Di, and Dehua Huang (National Center for Physical Acoustics, University, MS 38677)

The parabolic equation (PE) method has an advantage over normal-mode and fast-field program methods because it can easily handle a broad class of range-dependent problems. Normal-mode and fast field program methods, however, do provide useful information on the horizontal wavenumber spectrum, while the PE method gives only the total field. In some situations, such as upslope propagation, a spectral decomposition of the PE field into horizontal wavenumber components is a valuable tool for analyzing the acoustic field. This paper shows that for a PE field $p(r, z)$, the spectral decomposition $\phi(k, r, z)$ can be obtained by solving a one-dimensional elliptic equation. The derivation of the decomposition equation is given along with a numerical method for solving it. Numerical examples are given for several typical PE fields.

9:45

Z5. Effects of random-bottom topography on the parabolic-approximation determination of acoustic intensity. K. B. Mohrmann, M. J. Jacobson, and W. L. Siegmann (Rensselaer Polytechnic Institute, Troy, NY 12180-3590)

A model is presented for stochastic variations about a horizontal mean water-bottom interface below a shallow ocean channel. A pressure release surface and a rigid bottom are taken as boundary conditions for solutions of a parabolic approximation to the wave equation, when sound speed is depth dependent. A perturbation approach is used to generate a sequence of differential problems, the perturbation parameter being a measure of

the deviation in depth of the bottom. The mean and variance of statistics of the ratio of the mean to the variance of the ratio is treated, and the ratio is expressed in terms of the statistics of the random processes representing the bottom. Further, the effect on the statistics of the ratio is compared.

Z6. A self-consistent approach to wave propagation in weakly inhomogeneous media.

Program in Ocean Engineering, MA 02139) and George V. Frisk (Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

A unified perturbation approach to wave propagation and scattering in weakly inhomogeneous media is presented. By separating the total wave field into an averaged coherent propagation part and a random scattering part, a set of coupled wave equations is found. The problem of a wave propagating in a random medium is changed to a problem of a pair of coupled waves propagating in a deterministic medium with randomly distributed sources. This method not only improves upon the conventional Born approximation for the scattered field, but also gives the attenuation behavior of the coherent propagating field related to the statistics of the random media. Examples of underwater acoustic waves in an inhomogeneous water column and an inhomogeneous bottom are given.

10:15

Z7. Exponential growth of multipaths under chaotic conditions in a simple deterministic range-dependent waveguide. F. D. Tappert, M. G. Brown, and G. Goni (Rosensteil School of Marine and Atmospheric Science, University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149)

Propagation is considered in a simple range-dependent waveguide consisting of a constant sound-speed gradient, g , overlying a reflecting sinusoidal bottom. Let a and R denote the amplitude and wavelength of the bottom sinusoid. Previously it has been shown [F. D. Tappert and M. G. Brown, J. Acoust. Soc. Am. **83**, 537 (1988)] that when $\gamma = 4\pi^2 a c_0 / Rg$ exceeds 0.97 ray trajectories exhibit global chaos. Under such conditions, ray trajectories have an extreme sensitivity to initial conditions and environmental parameters; they are free to wander without bound in angle-depth space. Under such conditions it is shown that the number of eigenrays connecting a source and receiver on the bottom grows exponentially as the separation between them increases. Meanwhile, the average intensity of the eigenrays decays exponentially. The exponential growth and decay rates are compared to the average Lyapunov exponent for the ray trajectories. These results vividly demonstrate that under chaotic conditions predictability using ray theory is unattainable.

10:30

Z8. Mode coupling caused by a mesoscale oceanic eddy. E. C. Shang (CIRES, University of Colorado/NOAA/Wave Propagation Laboratory, Boulder, CO 80303)

From the full-wave theoretical point of view, there are two regimes for the interaction of acoustic waves with an oceanic mesoscale eddy: the weak coupling regime and the strong coupling regime. Understanding the limits of these two regimes is crucial for developing the new generation of ocean acoustic tomography systems based on the full-wave theoretical

framework instead of ray theory. In the present paper, the mode coupling caused by an oceanic mesoscale eddy is investigated by numerical simulation on CYBER 205. The eddy is modeled as a Gaussian eddy with parameters: *DR* (horizontal scale), *DZ* (vertical scale), *ZE* (center depth), and *DC* (strength). By incorporating a single-mode starting field into a 2-D PE (IFD) code, mode coupling coefficients are given by a mode filtering processing after passing through the eddy. Mode coupling coefficients are calculated for a frequency range of 10–100 Hz, and a mode number range of 1–10. It has been found that the mode coupling has a complex “resonant feature” with eddy parameters and also has a dependence on frequency as well as on mode number. However, the limit of the weak coupling regime, within which the adiabatic mode theory is valid, can be drawn out but it is by no means a simple criterion. [Work supported by NOAA.]

10:45

Z9. Effect of subbottom roughness on VLF acoustic propagation from a seamount. Stanley A. Chin-Bing (Naval Ocean Research and Development Activity, Numerical Modeling Division, Stennis Space Center, MS 39529-5004) and Joseph E. Murphy (Department of Physics, University of New Orleans, New Orleans, LA 70148)

Recently, two range-dependent ocean acoustic propagation and scattering models based on the finite-element (FE) method have been developed: the Finite-element Full-wave Range-dependent Acoustic Marching Element (FFRAME) mode and the hybrid finite element/parabolic equation PE-FFRAME model [J. E. Murphy and S. A. Chin-Bing, *J. Acoust. Soc. Am. Suppl. 1* **84**, S89–S90 (1988)]. These models have been applied to simulate the effects of subbottom roughness on the very-low-frequency (VLF) propagated acoustic field at both near- and long-range distances from a “seamount.” The “seamount” is simulated as an axisymmetric feature with gradual varying slope. Examples are presented that show the value of using these full-wave models (FFRAME and PE-FFRAME) over the usual one-way parabolic equation model when examining such underwater features. [Work supported by ONR and NORDA.]

11:00

Z10. A new split-step algorithm for solving acoustic propagation problems. David H. Berman, Evan B. Wright, and Ralph N. Baer (U.S. Naval Research Laboratory, Code 5160, Washington, DC 20375-5000)

A new equation is proposed to mimic the one-way aspects of the Helmholtz equation. This new equation allows a split-step solution like the conventional parabolic equation (PE) does, and thus it can be solved equally fast. Moreover, unlike the PE, the rays associated with high-frequency solutions of this equation are *exactly* the rays of the Helmholtz equation (even at high angles) for weakly range-dependent environments. For low-angle propagation, the equation reduces the conventional PE, and hence phases in this regime are given correctly. For what are now standard test cases, without severe bottom interaction, this new split-step algorithm performs as well or better than more complicated, and hence slower, one-way wave equation algorithms. [Work supported by ONR.]

11:15

Z11. Propagation modeling in a three-dimensional wedge using ray theory. Evan K. Westwood (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029)

A useful model for shallow-water, sand-bottom coastal regions is the three-dimensional (3-D) wedge. Unlike the 2-D wedge, source and re-

ceiver may lie *across* the slope from each other, rather than directly up-slope or down-slope. Acoustic propagation in the 3-D wedge is modeled using an extension of the complex ray method developed for the flat waveguide and the 2-D wedge [E. K. Westwood, *J. Acoust. Soc. Am. Suppl. 1* **83**, S79 (1988) and **84**, S90 (1988)]. However, due to the phenomenon of horizontal refraction (the bending of multiply reflected rays when viewed from above), each ray field in the 3-D wedge must be expressed as a plane wave integral over *two* angular variables, rather than just one. Evaluation of the integrals involves finding complex saddle points, each of which consists of a pair of angles giving the eigenray direction. For rays that are reflected near the critical angle, numerical integration along the path of steepest descent must be performed; otherwise, a first-order saddle point approximation may be used. The nature of the field in the 3-D wedge will be discussed, and possible benchmark problems will be suggested. [Work supported by ONR contract N00014-87-K-0346.]

11:30

Z12. The use of multipath time-delay estimation for source localization with passive broadband sonar systems. Peter Bilazarian and David J. Pistacchio (Raytheon Company, Submarine Signal Division, 1847 West Main Road, Portsmouth, RI 02871)

The employment of multipath time-delay estimation for acoustic source localization with passive autocorrelation and cross-correlation systems is examined. Important differences in multipath time-delay estimation with wideband and narrower band systems are delineated. Several algorithms are presented that convert multipath time-delay estimates into localization quantities, such as source depth or range. Some algorithms presume an isospeed medium, while another employs a procedure using the generic sonar model (GSM) and is applicable to a nonisospeed medium. Examples from a variety of multipath environments are provided that demonstrate consistent improvement in localization performance with the nonisospeed technique over the isospeed algorithms. Also, dramatic improvements are reported at ranges that exhibit a fairly complicated multipath structure. A theoretical comparison of the performance differences as a function of signal-to-noise ratio between a wavefront curvature ranging system and a multipath ranging system is provided. Finally, advantages of combining localization information obtained from wavefront curvature systems and systems which employ multipath time-delay estimation are discussed.

11:45

Z13. Coherent acoustic field computations using first-order multiple scattering and the generic sonar model. Susan M. Bates (MS 171, Raytheon Submarine Signal Division, 1847 West Main Road, Portsmouth, RI 02871-1087)

The generic sonar model (GSM) is modified to include, as an option, the computation of the coherent acoustic field in an ocean environment with internal waves. This modification to GSM permits estimation of coherent signal processing performance. GSM computes the acoustic pressure in a horizontally stratified ocean in the absence of environmental inhomogeneities using WKBJ solutions to the Helmholtz equation. Using the first-order multiple scattering approximation and the Garrett and Munk internal wave model, the coherent acoustic field is included in GSM for frequencies from 100 to 10 000 Hz. Results are presented using the modified GSM with internal waves. Comparisons are made with earlier results, which used the parabolic equation method [B. J. Bates and S. M. Bates, *J. Acoust. Soc. Am.* **82**, 2042–2050 (1987)], and GSM without internal waves.

Meeting of Accredited Standards Committee S2 on Mechanical Shock and Vibration

to be held jointly with the

Technical Advisory Group (TAG) Meeting for ISO/TC 108 Mechanical Vibration and Shock

S. I. Hayek, Chairman S2

Applied Research Laboratory, Penn State University, P. O. Box 30, State College, Pennsylvania 16801

D. F. Muster, Chairman, Technical Advisory Group for ISO/TC 108

4615 O'Meara Drive, Houston, Texas 77035

Standards Committee S2 on Mechanical Shock and Vibration. Working group chairs will present reports of their recent progress on writing and processing various shock and vibration standards. There will be a report on the interface of S2 activities with those of ISO/TC 108 (the Technical Advisory group for ISO/TC 108 consists of members of S2, S3, and other persons not necessarily members of those committees) including plans for the next meeting of ISO/TC 108, to be held in Milan, Italy in the spring of 1990.

WEDNESDAY MORNING, 24 MAY 1989

REGENCY B, 10:20 TO 11:55 A.M.

Session AA. Structural Acoustics and Vibration IV: Active Vibration Control

Gary H. Koopmann, Chairman

Department of Mechanical Engineering, Pennsylvania State University, University Park, Pennsylvania 16802

Chairman's Introduction—10:20

Contributed Papers**10:25**

AA1. Adaptive vibration control for structural mounting systems. Scott D. Sommerfeldt and Jiri Tichy (Graduate Program in Acoustics, The Pennsylvania State University, Applied Science Building, Box 30, State College, PA 16804)

It is often desired to isolate a source of structural vibration from surrounding structures. Typically, the vibrating structure is attached to the foundation via some form of passive, flexible mount, as a means of accomplishing the desired objective. One weakness common to all passive schemes is that they produce relatively little (or no) vibration reduction at the lower frequencies. Active control provides a means of compensating for this weakness to achieve good vibration isolation over a wide frequency range. In this study, adaptive closed-loop control has been combined with passive control to give both active reduction at the lower frequencies and passive reduction at the higher frequencies. The adaptive nature of the controller allows the controller to converge to the optimal solution, as well as to track any variations in the system parameters that may occur. A controller has been developed, based on the LMS algorithm, and investigated to determine its stability and convergence properties. As well, the mechanical coupling that occurs when multiple control actuators are used

has been studied. Real-time experimental results have been obtained from mounted vibrating structures, using a high-speed digital signal-processing board to run the control algorithm.

10:40

AA2. A method for controlling radiation from composite plates using embedded active fibers. Koorosh Naghshineh and Gary H. Koopmann (Noise Control Laboratory, 157 Hammond Building, The Pennsylvania State University, University Park, PA 16802)

A novel method of active control is proposed that focuses on controlling the acoustic radiation of a composite plate by directly altering its material properties. By integrating active elements within the plate, its material properties can be made to be time dependent and thus, given a fluctuating load, the material properties can be actively adjusted to produce a minimum radiation condition. The control method in this application can be best described as being quasiactive. In this study, a finite element model of a composite plate made of shape memory alloy fibers

and epoxy is used in conjunction with an acoustic radiation model to predict the effectiveness of using shape memory alloy fibers as a means of active noise control. Initial results demonstrate that the radiation characteristics of a simply supported plate comprising one side of an otherwise rigid rectangular box can be altered drastically by using the shape memory alloy fibers.

10:55

AA3. Optimization of placement of piezoelectric elements for the active control of plate vibrations. Janice I. Bayer, Vasundara V. Varadan, and Vijay K. Varadan (Department of Engineering Science and Mechanics, Research Center for the Engineering of Electronic and Acoustic Materials, The Pennsylvania State University, University Park, PA 16802)

Active vibration control of structures using piezoelectric sensors and actuators is explored. Finite element analysis providing theoretical predictions of the amplitudes of vibration of each of a mesh of nodes distributed over a plate is described. Statistical analysis of the predicted nodal amplitudes furnishes an optimized scheme for the placement of piezoelectric sensors and actuators. Said scheme provides for the minimum number of sensors and actuators needed to control the important vibration. Generalization of this algorithm to the control of other types of structures is discussed.

11:10

AA4. Further research on active control of flexural power flow in elastic thin beams. Luc O. Gonidou and Chris R. Fuller (Department of Mechanical Engineering, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

In a previous paper, preliminary results on active control of flexural power flow in elastic thin beams were presented [C. R. Fuller and L. O. Gonidou, *J. Acoust. Soc. Am. Suppl. 1* **84**, S47 (1988)]. The work described here will be concerned with further experimental and analytical work on the same subject. In particular, the control of flexural power flow from finite beams with various terminating impedances and different length beams is both analytically and experimentally studied and direct comparisons are now made. Parameters such as discontinuity impedance, bending nearfield generation, and number and location of control actuators and sensors are studied in light of the new test configurations. Mechanisms of control are discussed.

The use of a flexural power-based cost function is again demonstrated to be important in designing an efficient control strategy. [Work supported by NASA Langley Research Center.]

11:25

AA5. Large area sensors for active acoustic control systems. Thomas R. Howarth, Xiaoqi Q. Bao, Vijay K. Varadan, and Vasundara V. Varadan (Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802)

The design and development of a composite piezoelectric polymer (PVDF) sensor for use in an active acoustic control system are presented. Large area PVDF plates are encapsulated in a polymer matrix to detect both the incident and reflected signals in an underwater pulse tube at low frequencies. These signals are amplified, phase shifted, and then used to drive an absorbing transducer to negate reflections. The results of this system are compared with earlier data [X.-Q. Bao, V. K. Varadan, and V. V. Varadan, *J. Acoust. Soc. Am. Suppl. 1* **84**, S49 (1988)] that utilized piezoceramic spherical hydrophones. [Work supported by the Research Center for the Engineering of Electronic and Acoustic Materials.]

11:40

AA6. Experiments on optimization of active control on a three-dimensional acoustic source. Jeffrey A. Giordano, Kenneth A. Cunefare, and Gary Koopmann (Noise Control Laboratory, The Pennsylvania State University, University Park, PA 16802)

In this experiment, a box-shaped, acoustic radiator has top and bottom plates that are arbitrarily driven at their structural modes. The sides of the box consist of six loudspeakers that are used as active sources. An acoustic boundary element program is used as the basis for the active control strategy. The program, which allows active sources to be modeled at any position on the acoustic radiator, provides the magnitude and phase at which to drive the active sources to minimize the radiated power for a prescribed frequency and normal velocity distribution on the plates. The plate velocity profiles were determined via experimental modal analysis. The radiated power measurements, which were performed in a calibrated reverberation chamber, compared favorably to the theoretical predictions.

WEDNESDAY AM

Session BB. Physical Acoustics IV: General Topics

Michael R. Stinson, Chairman

Division of Physics, National Research Council, Ottawa, Ontario K1A 0R6, Canada

Contributed Papers

10:45

BB1. SH waves in an inhomogeneous elastic solid. Hyun S. Kim and J. H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

This paper concerns the propagation of *SH* waves in an inhomogeneous elastic half-space subject to a distributed shear excitation on the free-surface. The density and second-order elastic coefficient are assumed to deviate by small amounts from the average values, where such variations are periodic functions of the coordinate normal to the surface. Integral transform methods are used to solve the first- and second-order equations of motion, which are obtained from straightforward perturbation expansion of the displacement. Two cases of spatial distribution of shear excitation on the surface are considered; sinusoidal and Gaussian distribution. When the shear excitation is temporally periodic, it is shown that the periodic inhomogeneity can cause secular solutions. In the sinusoidal distribution case, either the first- or second-order displacement becomes secular if the wavelength of the signal and the periodicity of inhomogeneity and of the spatial distribution satisfy certain relations. In the case of a Gaussian distribution, it is shown that the second-order displacement becomes secular when the wavelength is twice the periodicity of inhomogeneity. [Work supported by NSF.]

11:00

BB2. Amplification of noise from liquid jets by entrained gas bubbles. H. S. Ribner (UTIAS, University of Toronto, Ontario M3H 5T6, Canada and NASA, Hampton, VA 23665)

This study reexamines turbulence-driven pulsations of gas bubbles in submerged liquid jets. The expectation of large amplifications—as much as 10^6 —of the jet noise under restricted conditions is confirmed. A macroscopic view via the dilatation theory of jet noise and a microscopic view examining individual bubble pulsations both predict sound power amplification equals (liquid sound speed/mixture sound speed).⁴ This follows also from the classical treatment of Crighton and Ffowcs Williams (1969). They related bubble monopole pulsations to Lighthill's jet noise quadrupoles in terms of scaling laws. Herein the bubble pulsations are related to those of local liquid monopoles by a direct amplification factor. This factor is quantitative rather than order-of-magnitude: It provides further insight into the properties of the bubbly jet noise. Strong inward refraction is expected when the bubbles are confined to the jet, leading to a focused axial lobe of enhanced emission. Both studies postulate a stable bubble population, but, in real flows, bubble breakup and the violent collapse of cavitation are prevalent and even noisier. Criteria for their occurrence are discussed.

11:15

BB3. Infrasonic emissions from atmospheric microbursts: Theory versus experiment. Allan J. Zuckerwar, Harlan K. Holmes, Jay C. Hardin (NASA Langley Research Center, MS 238, Hampton, VA 23665), and D. Stuart Pope (PRC Kentron, Inc., Hampton, VA 23556)

To demonstrate the existence of infrasonic emissions from atmospheric microbursts, and to investigate their properties in the frequency band 2–

20 Hz, a computational aeroacoustics technique has been applied to the simulated flow field of such an event. The theoretical development reveals that the emitted infrasonic pressure is proportional to the second time derivative of the vorticity within the flow. Thus peak pressure levels occur during the rapid deceleration and radial outflow developing as a microburst touches down upon the ground surface. The frequency spectrum falls at a rate of approximately 6 dB per decade. The computational results are compared to experimental records of naturally occurring infrasound taken throughout the year 1984, and support the point of view that in a certain class of these records the infrasound originated from atmospheric microbursts.

11:30

BB4. Acoustic waves over a transpiring surface. Yiping Ma, W. K. Van Moorhem, and R. W. Shorthill (Department of Mechanical Engineering, University of Utah, Salt Lake City, UT 84112)

An investigation of the effect on the acoustic waves due to wall mass injection in a piston-driven closed-end test chamber has been conducted. Dry ice is used as a source of mass injection. Acoustic pressure is measured with a pressure transducer mounted at the closed end. A comparison with the results with no side wall mass addition shows a significant increase in the acoustic pressure amplitude due to the interaction of the acoustic wave and side wall mass injection near resonance. A shift in the lowest resonant frequency between the two cases has also been detected. High-speed flow visualization and hot wire anemometry measurements all suggest that turbulence may occur in the acoustic boundary layer above the transpiring surface. This, in turn, feeds more energy into the acoustic wave. This increase in amplitude has never been reported before to the authors' best knowledge. [Work supported by U.S. Air Force Astronautics Laboratory and Morton Thiokol, Inc.]

11:45

BB5. At-sea measurements of underwater cavity resonance using a towed catamaran platform. S. A. Elder, L. G. Baker (U.S. Naval Academy, Annapolis, MD 21402), and S. Yoshikawa (5th Research Center, Technical Research and Development Institute, Japan Defense Agency, Yokosuka, 239 Japan)

Earlier investigations in the USNA high-speed tow tank, using a 4.5-ft towed model, had shown that large amplitude resonant oscillations can be produced by flow over a wall cavity immersed in water. For this case, it was found that critical speed is predictable from a Strouhal number formula derived for resonant cavity oscillations in air. Currently, an attempt to measure radiation from an underwater resonating cavity is being made, using the same towed model. Since "waveguide" cutoff of the tow tank is above oscillation frequency, it has been necessary to perform the measurements in deeper water. For this purpose a 17-ft catamaran has been adapted for use as a towing platform to make it possible to take measurements in Chesapeake Bay. A YP class vessel, equipped for oceanographic research, is used for towing and for recording data. For speeds up to 10 kn, f_{11} and f_{12} modes have been detected at the expected speeds. Current results for amplitude and radiation will be described.

Session CC. Engineering Acoustics V: Materials' Properties Evaluation

Thomas R. Howarth, Chairman

*Department of Engineering Science and Mechanics, Pennsylvania State University, University Park, Pennsylvania 16802***Chairman's Introduction—12:50*****Invited Papers*****12:55****CC1. Characterization and development of transducer elastomers.** Robert Y. Ting (Underwater Sound Reference Detachment, U. S. Naval Research Laboratory, Orlando, FL 32856-8337)

The failure of underwater transducers is often related to problems with the materials used, such as elastomers, ceramics, adhesives, coatings, and fill fluids. The general practice has been to select proprietary materials from different commercial sources, and use them as they are available for the fabrication of a transducer. Such practices allow the designer very little control of the quality of the materials. The result is the production of devices that are doomed to fail because they are nonreproducible, or contain materials incompatible to each other. In recent years, improper choices of elastomer have been typical in causing large numbers of failure in sonar transducers. Research efforts at NRL-USRD have therefore been focused on the development of composition-property relationships for elastomers by using well-characterized samples. Instrumentational techniques were developed for the analysis of chemical and acoustical properties. These techniques were subsequently used for materials evaluation and quality control. Based on these research results, specialty elastomers were also successfully developed for specific transducer applications. [Work supported by ONT and NAVSEA.]

1:20**CC2. Characterization of viscoelastic materials used to control underwater sound.** Wayne T. Reader (Vector Research Company, Inc., Suite 1200, 6903 Rockledge Drive, Bethesda, MD 20817)

The control of sound fields within confined regions such as water-filled tanks may require the use of a combination of three types of materials: one designed to absorb waterborne sound, one designed to block the passage of waterborne sound, and one that damps structural vibrations. The most frequently used materials consist of viscoelastic polymers containing voids or gas filled inclusions in volume fractions that are essentially zero for damping materials, are of the order 10% for absorbers, and that may approach 90% for blocking materials. The empirical characterization of these very different materials during their development or prior to their installation is achieved with a variety of apparatus and techniques. Typical forms of these apparatus, the associated principles of operation, the physical properties that are measured, and limitations of the techniques will be reviewed. Described will be procedures for obtaining the most fundamental property of underwater acoustical materials, viz., the complex modulus, the use of water-filled waveguides to determine the reflectivity and transmissivity of small samples at normal incidence, and measurement of the same properties on finite sized panels at oblique angles. Finally, contrary to common wisdom, a procedure is proposed for "scaling" acoustical materials.

1:45**CC3. Characterization and analysis of viscoelastic materials over wide frequency and temperature intervals.** W. M. Madigosky (NSWC, 10901 New Hampshire Avenue, Silver Spring, MD 20903)

Typically, the dynamic behavior of materials is a strong function of both temperature and frequency. An accurate knowledge of the dynamic properties over a wide interval of temperature and frequency is, therefore, essential to the efficient design and engineering use of the viscoelastic phenomenon. Recently, both laboratory and commercial apparatus for making measurements over the required large frequency and temperature ranges have become available. This paper reviews the current status of the measurement techniques, the use of the WLF time temperature shift equation, and presents data on a number of pure materials and polymer blends that illustrate the capabilities currently available. Finally, the data may be analyzed in terms of the fractional operator model that describes both the storage and loss moduli with a single complex equation. The adequacy of the model to fit the data will be discussed.

CC4. Characterization of piezoelectric ceramics for high-power transducers. Paul Gonnard and Lucien Eyraud (Lab. de Génie Electrique et Ferroélectricité, Ins. Nat. des Sciences Appliquées, 69621 Villeurbanne, France)

The choice of piezoelectric materials for a high-power electromechanical conversion must take into account various factors such as the figure of merit of the material (FOM), the stability of the electromechanical parameters, and the dielectric and mechanical losses. Materials suitable for impact-type igniters must present both a high FOM and a good stability under repetitive shocks. For high-power electromechanical conversion, the FOM of a single piezoelectric bar and of a Langevin-type transducer depends on the ratio of the piezoelectric coefficient d_{33}^t over the elastic compliance s_{33}^t of the material. Without losses, the limiting factor would be the maximum dynamic stress at a vibration mode. Methods for characterizing the dielectric and mechanical losses under high driving level are given. The electromechanical characteristics and losses of some new compositions are compared with those of standard "hard" materials. An interpretation of the high-level behavior is given taking into account a compensation of the oxidation state of manganese during cooling by a change in the valency of the substitution ions. [Work supported by DCAN.]

CC5. Evaluation of the property coefficients of piezoelectric polymers. Donald Ricketts (Raytheon Company, Submarine Signal Division, Portsmouth, RI 02871-1087)

Analytical models are presented for evaluating the property coefficients of thick-film piezoelectric polymers. In particular, these models facilitate the evaluation of the complex elastic compliance coefficients $\hat{s}_{11}^D, \hat{s}_{12}^D, \hat{s}_{22}^D, \hat{s}_{33}^D$, and \hat{s}_{66}^D from the measured set of quantities $\{f_m, f_i, f_n, A_i, A_n\}$, where f_m is the resonance frequency of the test specimen $f_i < f_m < f_n$ and A_i and A_n are the relative amplitudes at the selected frequencies f_i and f_n . Impact excitation of the test samples, a vibration sensor, and a FFT-based spectrum analyzer are used to measure these quantities. In addition, a model is presented for evaluating the complex electromechanical coupling factor \hat{k}_{31} from the measured value of the transfer function of the length expander polymer bar with divided electrodes. Experimental results were obtained on PVF₂ samples made by Raytheon Research Division, and the analytical models were used to evaluate the property coefficients of these samples. The results of these evaluations for $\hat{s}_{11}^D, \hat{s}_{11}^{D*}, \hat{s}_{12}^D, \hat{s}_{22}^D, \hat{s}_{33}^D, \hat{s}_{66}^D$, and \hat{k}_{31} are presented.

WEDNESDAY AFTERNOON, 24 MAY 1989

AMPHITHEATER, 1:00 TO 3:00 P.M.

Session DD. Architectural Acoustics II: Room Acoustics, Laboratory Measurements, Etc.

Angelo J. Campanella, Chairman

Campanella Associates, 3201 Ridgewood Drive, Columbus, Ohio 43026

Contributed Papers

1:00

DD1. Ambient noise in open plan offices. J. B. Moreland (Westinghouse Research and Development Center, 1310 Beulah Road, Pittsburgh, PA 15235)

This paper summarizes ambient noise data that were collected in open plan offices over the past 2 years. The approach centered on monitoring the noise level in empty workstations to establish some statistics of the noise that would be auditioned by a person sitting in that workstation during a normal work day. Offices both with and without electronic sound masking were investigated, as were offices with either acoustically hard or acoustically soft screens. It was found that the noise level in an individual workstation increases with the number of people in this office. For the 25 workstations considered in this investigation, the average daily noise level in a workstation ranged from 42.2–58.6 dBA. The noisier workstations

usually had both (i) electronic sound masking and (ii) acoustically hard screens, whereas the opposite was usually found for the quieter workstations.

1:15

DD2. Propagation and reverberant phase in acoustical spaces. Lan Liu and Richard H. Lyon (Department of Mechanical Engineering, MIT, Cambridge, MA 02139)

Systems with low modal overlap cannot support a "direct field" and have also been shown to have a phase trend and corresponding group delay that is far in excess of the expected propagation delay. The phase trend that is associated with this behavior has been termed "reverberant phase." The reverberant phase also shows up in systems with high modal

overlap, but in such systems, the phase of the direct (or propagation) field is expected to also be present. If the effects of reverberation could somehow be subtracted from the data, one would expect to see the propagation phase. In this paper, a procedure for the decomposition of propagation and reverberant phase components from a received signal is presented, using properties of the complex cepstrum. These ideas are tested experimentally using data from transfer functions for small acoustical chambers. Directions for future work on this topic will also be discussed. [Work supported by NTT Research and Development Center.]

1:30

DD3. Measuring and using the impulse response of a room. E. Paul Palmer and David A. Berry (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602)

The point-to-point impulse response of a room is a record of the impulsive sound from a single source to a single microphone. Different systems and signal-processing methods may be used in this measurement depending on the accuracy required. If only the time history of major echoes is desired in order to trace (and correct) reflecting surfaces, the energy-time curve of a time-delay spectrometer system is excellent. An *m*-sequence signal, with its speed of processing, is versatile for a homebuilt PC-based system and comes closer to a true impulse response. For an impulse response adequate for convolution with a signal, in order to accurately simulate a room, more care is required. The poor frequency response and directionality of the loudspeaker must be nullified to reduce it to a "perfect" point source. A computer-based system to approach this ideal will be described. Further, a real source might be an orchestra rather than a point, and the usual receiver has two ears. Research to graphically display a binaurally recorded impulse response and to deconvolve earphone characteristics (as was done with the loudspeaker) will be reported.

1:45

DD4. The role of the microphone in parametrizing the acoustics of three rectangular concert spaces. J. M. Mastracco (School of Engineering, Rensselaer Polytechnic Institute, Troy, NY 12181 and WRPI-FM 91.5, Troy, NY 12181)

The scientific method is a principle that states that no theory or model of nature is tenable, unless the results it predicts are in accord with experiment. The concert hall, as a physical system, has long been misunderstood in terms of the effects it has on acoustics. This paper discusses the application of the scientific method to the measurement of acoustical parameters in three renowned spaces: Boston Symphony Hall, the Troy Music Hall, and the Union College Memorial Chapel. The results of the experiments strongly suggest that no characterization of a hall's acoustics is possible without the specification of the microphone used in the measurement. Microphones ranging from a 1/2-in. omnidirectional transducer to an acoustic manikin employing a 1/2-in. omnidirectional transducer produce clearly identifiable differences in reverberation times and other widely used parameters that are calculated from the transient response of the space. These results suggest, contrary to the belief of many, that acoustical science applied to concert halls is not an *exact science*, but pathological in character. It is concluded that acoustical comparisons between different spaces are not possible unless the type of microphone, as well as its placement and orientation in the space, are specified. [Work supported in part by the Audio Engineering Society Education Foundation.]

2:00

DD5. Computer graphics for acoustic simulation and visualization. Adam Stettner (Program of Computer Graphics, Cornell University, Ithaca, NY 14853)

Computer simulation can be used to generate the spatial and temporal data describing acoustical behavior. By using computer graphics to display the multidimensional data, substantially greater amounts of information than conveyed by standard techniques can be communicated to the designer. A methodology is presented for evaluating an acoustical environment using a two phase approach. The first phase entails simulating the time varying spatial distribution of sound energy, using techniques similar to those used in computer graphics to model illumination. These techniques, specifically probabilistic geometric ray tracing and reflection modeling, have been highly refined. By modifying these methods to account for a few basic differences between light and sound, they can be well suited to the simulation of sound propagation. The second phase entails an investigation of techniques of displaying multidimensional data that are useful to the user. These techniques, which include the use of three-dimensional color image generation, animation, and icon representation, are used to visualize acoustical measurements derived from the simulation data. Using these criteria, three variations in the design of a performance hall are evaluated and compared. [Work supported by NSF.]

2:15

DD6. Standard measurement of sound transmission through suspended ceilings. J. D. Quirt and R. E. Halliwell (Acoustics Section, Institute for Research in Construction, National Research Council of Canada, Ottawa, Ontario K1A 0R6, Canada)

Sound transmission through the space above the suspended ceiling commonly limits insulation between offices where interoffice partitions do not extend above the suspended ceiling. This paper presents a detailed comparison of standard test methods for evaluating suspended ceiling performance. A laboratory has been constructed that can readily be converted to satisfy requirements of ISO Standard 140 Part 9, or a corresponding ASTM draft standard, or the Acoustical Manufacturers' Association standard (AMA I-II) from which these descended. Characteristics of the test environment have been studied, and a series of suspended ceiling systems have been tested using all three standard methods. Significant differences and their causes will be discussed.

2:30

DD7. Sound transmission measurements through concrete block walls with attached drywall layers. A. C. C. Warnock (Institute for Research in Construction, National Research Council, M27 Montreal Road, Ottawa, Ontario K1A 0R6, Canada)

Sound transmission through a series of concrete block walls was measured in a reverberation room according to ASTM E90. The measurement frequency range was extended down to 63 Hz. The effects of the method of attaching the drywall and of adding sound absorbing material to the cavity were investigated during the series. The series produced a reasonably well-behaved set of data even at the lower frequencies. As expected, increasing the depth of the cavity and adding sound absorbing material moved the mass-air-mass resonance to lower frequencies and usually improved the sound transmission class rating. The highest sound transmission class measured was 73. Because of the extended range of the measurements, reductions in the low-frequency transmission loss values were quite evident. A simple method of predicting the results obtained will be presented and its extension to other cases discussed.

2:45

DD8. Transfer function method for measuring characteristic impedance and propagation constant of porous material. Hideo Utsuno (Department of Mechanical Engineering, University of Kentucky, Lexington, KY 40506-0046 and Applied Mechanics Center, Kobe Steel, Ltd., Takatsukadai 1-chome Nishi-ku, Kobe, 673-02, Japan), Toshimitsu Tanaka, Takeshi Fujikawa (Applied Mechanics Center,

A method for measuring the characteristic impedance and propagation constant of porous materials is described. Measurements were performed based on a surface impedance method that required a set of distinct acoustic impedances derived at the material surface. This requirement is satisfied by arbitrarily changing the air space depth behind the material, and then a new formulation is derived so that a recently

developed method of determination, called the transfer function method, can be applied. An appropriate set of air space depths is also discussed. Glasswool and porous aluminum were used to assess the usefulness of the present method. The normal acoustic impedance and normal absorption coefficient of the test materials with arbitrary thicknesses or with an arbitrary air space depth were calculated from the obtained characteristic impedance and propagation constant, then they were compared with the measured values that were obtained directly using the transfer function method. The good agreement achieved suggests that the present method is reliable and effective enough to measure the characteristic impedance and propagation constant over a broadband frequency range.

WEDNESDAY AFTERNOON, 24 MAY 1989

REGENCY B, 1:00 TO 3:00 P.M.

Session EE. Musical Acoustics III: General Topics

Donald E. Hall, Chairman

Physics Department, California State University, 6000 J Street, Sacramento, California 95819

Contributed Papers

1:00

EE1. Efficient real-time musical spectrum analyzer using DSP chip. William F. McGee (Department of Electrical Engineering, University of Ottawa, Ottawa, Ontario K1N 6N5, Canada)

A 72-note transcriber for polyphonic music has been realized. A Texas Instruments TMS320C25 digital signal processor operating with a 40-MHz clock (100-ns instruction times) has been programmed to provide a musical spectrum analyzer. The analyzer is a bank of 72 filters, tuned to the frequencies of western music. Each filter has a transfer function of unity at its center frequency and a zero at the 71 other frequencies. Due to the design, only 14 instructions per filter per sample are needed. Logarithmic response of each filter is computed at a slower rate, once every 72 samples. Response times appear adequate to permit transcription of polyphonic music. Postprocessing to remove weaker notes a semitone away is found to improve the transcription capabilities, whereas removing partials does not appear to offer as much improvement. The transcriber is tested by passing the output to a MIDI synthesizer. Synthetic music is reasonably accurate, other music is usually recognizable; further postprocessing will improve performance. [Work supported by BNR and NSERC.]

1:15

EE2. Computer simulation of the guitar, work in progress. Harold A. Zintel (Acoustics Department, Pennsylvania State University, State College, PA 16801)

The guitar without strings is modeled by the displacement response of the bridge and the pressure response at an observer location due to an impulse forcing function at the bridge. The model thus entails nine unique impulse response functions, six for combinations of impulse directions and displacement responses and three for the observer location pressure due to each direction of the impulse direction. These nine impulse responses are obtained experimentally. A finite difference approach is used to model a string and its interaction with the instrument. The observer pressure levels predicted by the program are plotted as a waterfall spectrum and also sent to headphones by way of a D/A conversion. By conducting modal analysis of the guitar, the effect of modifications to the structure of the instrument can be estimated. Further, analyzing the new modal parameters using an acoustic boundary element method (CHI), the impulse pressure response can be estimated. The final result, when the project is complete, is a tool that can be used to predict the sound of a particular instrument after structural modifications.

EE3. Statistical energy analysis modeling and testing of an acoustic guitar. Daniel J. Hosek and Glen E. Johnson (Department of Mechanical Engineering, Box 8-B, Vanderbilt University, Nashville, TN 37235)

Principals of statistical energy analysis (SEA) were applied to develop a mathematical expression for the time-averaged, space-averaged energy of vibration in specified frequency bands for the top plate of an acoustic guitar. The model predicted energy in terms of string parameters, guitar parameters, and plucking data. Primary assumptions were that the damping of the strings was negligible, that energy flowed from the strings to the resonance box, and that energy was dissipated in the resonance box. Power supplied to the strings was estimated from a classical model of the vibrating string. An experimental plucking apparatus was designed and instrumented with Bruel & Kjaer vibration analysis equipment. The results of the SEA-based computer program were compared to actual experimental data for a Yamaha FG335 acoustic guitar. The SEA model tended to overpredict the energy in the top plate in most frequency bands. Possible causes for the differences were identified and discussed.

EE4. Statistical data modeling approach for determining the acoustic impedance of musical instruments. Gerald J. Lemay (Electrical and Computer Engineering Department, Southeastern Massachusetts University, North Dartmouth, MA 02747)

Recent theoretical models [M. E. McIntyre *et al.*, J. Acoust. Soc. Am. 74, 1325-1345 (1983)] for the oscillation mechanism of musical instruments have utilized an analytic closed-form relationship between the Green's function $g(t)$, the reflection function $r(t)$, and the acoustic impedance Z_0 (which may also be time varying). Given two of the three quantities $\{g(t), r(t), Z_0\}$, the method used to find the third function usually involves the discrete Fourier transform. This paper presents an alternate approach to the problem of estimating Z_0 from measurements of $g(t)$ and $r(t)$. Statistical data modeling in the context of digital filter design provides a solution that does not require the computation of a discrete Fourier transform. It is shown that the acoustic impedance is the impulse response of the cascade of two digital filters whose coefficients minimize the fitting error between an (M, N) th order model and the observed sequences.

EE5. Model-based trumpet tone synthesis. Paul H. Dietz (Department of Electrical and Computer Engineering, Carnegie Mellon University, Pittsburgh, PA 15213)

In an effort to produce realistic trumpet tones, a detailed model of the dynamics of the instrument and the player was created. The trumpet was modeled as a collection of linear circuit elements to account for the effects of the mouthpiece, tubing, bell, and radiation impedance. Testing was performed using both impulse response and equivalent length measurements, verifying the accuracy of the model. The model of the player extends from the lungs through the lips and contains both linear and nonlinear elements. The two nonlinearities are due to the inelastic collision of the lips, and the impedance of the lip opening. These models were computer simulated varying three control parameters: the spring constant of the upper lip, the force keeping the lips closed (which can also be thought of as an offset in the spring's zero position), and the pressure in the lung. Listeners report that the resulting simulated tones are quite realistic. These tones demonstrate many of the attributes of real trumpet tones, including the tendency to lock onto the same notes in the harmonic series.

EE6. Musical pitch tracking based on a pattern recognition algorithm. Judith C. Brown (Media Laboratory, MIT, Cambridge, MA 02139 and Department of Physics, Wellesley College, Wellesley, MA 02181)

It can be shown that, when the Fourier transform is plotted against log frequency, the pattern obtained in the frequency domain is the same for any sound with harmonic frequency components. A constant Q Fourier transform has been calculated for frequencies corresponding to a musical quarter tone and plotted against log frequency. The resulting pattern can be "tracked" using cross correlation to give the musical pitch. Excellent results have been obtained for a variety of instruments including the violin, flute, and piano. [Work supported by a Brachman Hoffman Fellowship from Wellesley College.]

EE7. Effects of preceding scale on melodic pitch interval judgment. Minoru Tsuzaki (ATR Auditory and Visual Perception Research Laboratories, Inuidani, Seika-cho, Kyoto 619-02, Japan)

To investigate interactive aspects of musical scale systems and pitch intervals in human musical cognition, the melodic pitch interval was judged in three preceding scale contexts: (a) diatonic, (b) chromatic, and (c) no scale. Subjects compared standard and comparison pitch intervals using three response categories: "small," "equal," and "large." Standard intervals began with note B, or note C, and ascended by 100, 150, or 200 cents. A "start note effect" is the most prominent for the 200-cent standard in the diatonic context. When it began with note B, i.e., a leading tone in a diatonic context, a peak in the "equal" response histogram disappeared, and the boundary between "small" and "large" response moved far away from its musically equal point. For the 100-cent standard, judgments were rather more precise in the C start condition than in the B start condition, irrespectively of the scale context. For the 150-cent standard, a "start note effect" was not observed clearly in any scale context. Detailed discussion is provided on the role of the diatonic scale system as a referential frame in view of two aspects of pitch interval judgement, that is, size and equality relationship.

EE8. Adults' perception of Western and Javanese musical scales. Michael P. Lynch, Rebecca E. Eilers, and D. Kimbrough Oller (Departments of Psychology and Pediatrics, University of Miami, P. O. Box 016820 (D-820), Miami, FL 33101)

A cognitive/perceptual model of the processing of musical intervals was investigated by assessing the role of experience with Western music in perception of mistunings in both Western and non-Western scales. Musically inexperienced subjects, persons with limited background in music, and professional musicians heard seven-note major, minor, and Javanese pelog melodies. Each of the melodies contained successively the first, second, third, fourth, fifth, third, and first pitches of each scale, with the first and last pitches always D4 (293.66 Hz). The frequency relationships of the pelog scale were maintained within this framework. Each melody was altered in two ways. First, the top pitch (A4, 440 Hz in major and minor; 461.3 Hz in pelog) was raised along a continuum in ten 0.4% steps. Second, any pitch but the first was randomly selected for sharpening at each of ten 0.5% steps. PEST was used to estimate the smallest degree of pitch raising on these continua that subjects could reliably perceive for each scale type. As predicted by the model, results indicate a strong relationship between musical training and perception of tuning alterations in the Javanese scale. Musically sophisticated subjects had equal perceptual thresholds across the three scale types. Musically naive subjects had a perceptual advantage for Western scales. Patterns of subject responding differed slightly when pitches were randomly selected for sharpening as opposed to when only the top pitch was raised. [Work supported by Am. Psychol. Assoc.]

Session FF. Noise V: Noise Control and Effects (Poster Session)

M. G. Prasad, Chairman

*Department of Mechanical Engineering, Stevens Institute of Technology, Castle Point, Hoboken, New Jersey 07030***Contributed Papers**

All posters will be on display from 1:00 to 3:00 p.m. (Viewing hours will also be extended into the evening and through noon on Thursday.) To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 to 2:00 p.m. and contributors of even-numbered papers will be at their posters from 2:15 to 3:00 p.m. Contributors are encouraged to leave their posters in place until 12:00 noon on Thursday. A cash bar will be set up in the evening to facilitate informal discussion. The Goldstein Auditorium will be closed at 10:00 p.m. and reopen on Thursday morning.

FF1. Application of an acoustic noise synthesis technique in the design of a duct system. M. G. Prasad (Department of Mechanical Engineering, Stevens Institute of Technology, Castle Point, Hoboken, NJ 07030) and T. V. Ananthapadmanabha (AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974)

This paper discusses the application of acoustic noise synthesis (ANS) for the design of a duct system, specifically, a muffler system. ANS is based on well-known principles of speech synthesis using a source-filter model. The designer can use an ANS technique to study the influence of design parameters of the duct system on the perceptual quality of noise. This approach is useful in studies relating to perceptual loudness criteria and noise control. The model considered in this study is a plane-wave white Gaussian noise source, a simple expansion chamber-type muffler, and an open-end tail pipe. The system transfer function is obtained based on the transmission coefficient expression derived from the chain matrix. Impulse response of the system is obtained by the application of an inverse FFT. The impulse response is convolved with the source signal to obtain the system output in the time domain. The output fed through a digital-to-analog converter and a loudspeaker yields the simulated acoustic noise. This simulation procedure can be applied for any given model for source-muffler termination. The advantage of acoustic noise synthesis is that the source-muffler interaction, muffler design changes, and the effects of boundary conditions can be simulated based on theoretical models.

FF2. Noise control in microelectronics clean rooms. Colin G. Gordon and Amir M. Yazdannyaz (BBN Systems and Technologies Corporation, 21120 Vanowen Street, Canoga Park, CA 91303)

Modern clean rooms of the sort used by the microelectronics industry may entail 600 air changes per hour. They pose, therefore, unique noise control problems. These problems are exacerbated by the fact that conventional acoustically absorbent materials cannot be used within the clean room. In this paper, some typical sound data that have been measured in clean rooms are presented. Also discussed are the noise criteria that are desirable for personnel working within the space and for sound-sensitive equipment that may occupy the space. Some practical ideas for noise control are suggested.

FF3. Duct attenuation with bulk reacting lining. David A. Bies and Colin H. Hansen (Department of Mechanical Engineering, University of Adelaide, GPO Box 498, S.A. 5001, Australia)

Attenuation of lowest-order modes in lined straight ducts of both rectangular and circular cross section is analytically and experimentally in-

vestigated. The lining is assumed to be anisotropic, bulk reacting, and covered by a thin impervious membrane. The effect of flow for low Mach numbers is also taken into account. Frequency averaging within octave bands is used to produce useful design curves. Limited comparison between prediction and measurement will be reviewed and is encouraging.

FF4. Physiologic effects of environmental noise and talking in coronary care patients. Carol F. Baker (College of Nursing, The Ohio State University, 1585 Neil Avenue, Columbus, OH 43210)

The purpose of the study was to describe changes in heart rate and blood pressure in patients with acute myocardial disease when exposed to noise and talking in the environment. A convenience sample of ten adult patients admitted to an 18-bed university hospital CCU were selected as subjects. Private rooms separated patients from an open nurses' station. Sound-pressure level and content of sounds were measured with a Bruel & Kjaer type 2230 sound level meter; the microphone was placed above the head of the bed. Heart rate and ECG from the bedside monitor were recorded simultaneously with dBA and sound on a TEAC R-61 physiologic tape recorder and later printed onto a 2600S Gould chart recorder. Blood pressure was measured every 5 min with a Dinamap. Data collection occurred during four 45-min periods on two consecutive days. Mean sound-pressure level of noise sources were: toilet flush—76 dBA, alarms—66 dBA, talking inside the room—65 dBA, and ambient—49 dBA. Repeated measures ANOVAs compared noise sources, content of conversation, and roles of people with changes in decibel, heart rate, and blood pressure. Short bursts of noise showed the greatest mean change in maximum dBA, followed by talking of nurses or doctors inside the room. Heart rate changes were greater when family talked. Mean arterial pressure was higher during talking and background noise. [Work supported by a University Seed Grant.]

FF5. Efficiency of absorbent layers on barrier. André L'Espérance (Groupe d'Acoustique de l'Université de Sherbrooke, Département de Génie Mécanique, Université de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada)

This paper presents a method for calculating the insertion loss of barriers having the source side, the receiver side, or both sides covered with an absorbent material. In this method, the approximate solutions for diffraction over an absorbent wedge proposed by Koers is combined with a classical model of sound propagation over an impedance ground. Theoretical and experimental results are compared for various geometrical configurations and barrier surface conditions and confirm the accuracy of

the method proposed. This study has enabled one to discover that it is more efficient to place the absorbent covering on the surface of the barrier associated with the most important angles of diffraction (source top edge and/or receiver top edges). When these angles are about the same on both

sides of the barrier, an absorbent covering will reduce the diffracted field, no matter if it is placed on the source side or on the receiver side. These results explained various conclusions found in literature about the usefulness of absorbent barriers compared to hard barriers.

WEDNESDAY AFTERNOON, 24 MAY 1989

REGENCY A, 1:00 TO 3:00 P.M.

Session GG. Physical Acoustics V: Relaxation, Sonic Booms, and Propagation

Richard Raspet, Chairman

Department of Physics and Astronomy, University of Mississippi, University, Mississippi 38677

Contributed Papers

1:00

GG1. Propagation of sound in vibrationally excited N_2/CO and CO/H_2 mixtures. Timothy H. Ruppel and F. Douglas Shields (Department of Physics and Astronomy, The University of Mississippi, and the National Center for Physical Acoustics, University, MS 38677)

Measurements of the resonant reverberation of sound in a tube containing N_2/H_2 , N_2/He , N_2/CH_4 , and N_2/H_2O mixtures have shown an amplification of the sound following a rapid excitation of the gas by an electric discharge [F. D. Shields and L. D. Lafleur, *J. Acoust. Soc. Am.* **83**, 2186 (1988)]. This effect has been named SACER (sound amplification from controlled excitation reactions). This paper reports similar measurements in N_2/CO mixtures, and also reports preliminary results from measurements in CO/H_2 mixtures. It has been possible in the past to determine vibrational relaxation times and relaxation times for the conduction of translational and vibrational energy to the tube wall from the changes in the translational temperature in the gas following the electric discharge. In the work reported here, these rates have been confirmed by direct measurement of the decay times of the CO infrared emission. [Work supported by ONR.]

1:15

GG2. Acoustic absorption in high-temperature gas mixtures of atmospheric constituents. Wallace George (Noise Control Laboratory, Pennsylvania State University, University Park, PA 16802)

Experimental data on acoustic absorption at high temperatures (up to 1000 K) due to vibrational molecular relaxation are presented for gas mixtures of O_2 , N_2 , CO_2 , and H_2O . These absorption tests represent some of the first acoustical measurements in gas mixtures at these elevated temperatures. These data are compared with the predicted results of the matrix formalism for sound absorption in multicomponent gas mixtures, due to molecular relaxation. This formalism, which is based on irreversible thermodynamics, was utilized by Bauer, Bass, and others. The literature was examined to determine the temperature dependence of the various empirically derived energy transfer rates required by this model. Some of the rates were adjusted to correspond with the current data. This means that the empirical rate constants are adjusted based on high-temperature acoustical tests, rather than on less reliable shock tube data.

1:30

GG3. Propagation of nonlinear transients, such as sonic booms, through an absorbing and relaxing atmosphere. Jongmin Kang and Allan D. Pierce (Noise Control Laboratory and Graduate Program in Acoustics, Pennsylvania State University, 157 Hammond Building, State College, PA 16804)

Recent resurgence of interest in sonic boom propagation through the atmosphere necessitates detailed understanding of how absorption, relaxation, ray focusing and defocusing, and nonlinear steepening interact. Previously used techniques, such as by Bass *et al.* (1983) for simultaneously treating molecular relaxation and nonlinear propagation, although yielding important insights, have required extensive computation time and have been limited to simple source models. The present paper gives the extension of Burger's equation, with relaxation included, to propagation through inhomogeneous atmospheres using the wedding of nonlinear propagation and ray acoustics suggested by Hayes. It is argued that, in the absence of shock formation, and for lower amplitude waveforms, relatively large step sizes may be used with nonlinear steepening, dispersion, and absorption treated as additive effects, the former being easily handled using the implicit exact solution of the inviscid Burger's equation that incorporates the concept of an age variable. After something that resembles a shock appears, nonlinear effects, relaxation, and absorption must all be simultaneously considered in the computation of waveform details over the narrow risetime of the shock, but a perturbation technique based on the method of multiple scales shows promise of cutting down the computation time. [Work supported by NASA-LRC and by the William E. Leonhard endowment to Pennsylvania State University.]

1:45

GG4. Extended Huygens-Kirchhoff method for predicting transient sound propagation through inhomogeneous moving media. Spiro Kouzoupis and Allan D. Pierce (Graduate Program in Acoustics, Pennsylvania State University, 157 Hammond Building, State College, PA 16804)

The linear acoustics equations for when the ambient medium has inhomogeneous flow yield an integral corollary analogous to that originally derived by Helmholtz for the reduced wave equation and by Kirchhoff for

the wave equation. Such a corollary involves a Green's function for the ambient medium that is here approximated using the eikonal approximation following the technique of Sommerfeld and Runge. The corollary enables one to predict transient waveforms at distant points from information prescribed over a surface and improves upon ordinary geometrical acoustics in that it can account for the presence of caustics in the intervening space. Huygens' principle is realized by this corollary and geometrical acoustics and parabolic equation approximations arise as limiting cases. Numerical examples are given for some simple cases of flows and the use of the overall technique to explore sonic boom propagation in the turbulent atmosphere is outlined. Although the Green's function is approximate, the relation is a good approximate technique for advancing the overall field over moderate distances, without having to explicitly consider the field at intermediate points. [Work supported by NASA-LRC and by the William E. Leonhard endowment to Pennsylvania State University.]

2:00

GG5. Anomalies in backscattering on large-scale inhomogeneities. Yevgeny G. Schemelev (Department of Radiophysics, Gorky State University, Gorky, USSR and Sibley School of Mechanical and Aerospace Engineering, Cornell University, Ithaca, NY 14853)

The backscattered field is investigated under conditions such that the plane monochromatic wave falls on a layer of chaotic statistically isotropic inhomogeneities. It is shown that the main part of the received signal comes from a specific direction α (determined by geometry and scale), if the backscattering signal is received by a directed antenna and the condition $d < l$ is realized (d is the receiving signal aperture, l is the characteristic scale of the inhomogeneities). This effect may be of use for determination of characteristic scales of large-scale inhomogeneities from analysis of the back-scattering signal.

2:15

GG6. Laser-Doppler vibrometry measurements of acoustic-to-seismic coupling and geophone-ground coupling ratios. W. Pat Arnott and James M. Sabatier (National Center for Physical Acoustics, P. O. Box 847, University, MS 38677)

The surface of the ground is an air-filled porous material because of weathering and vegetation. Typically, soils are porous to a depth of several feet. An airborne sound wave can couple into the ground as a pore fluid wave. The pore fluid wave is attenuated through viscous drag at the pore walls; hence, energy can be coupled into the soil matrix as seismic motion [J. M. Sabatier *et al.*, *J. Acoust. Soc. Am.* **79**, 1345-1352 (1986)]. This mechanism gives a coupling ten times that which would be expected for momentum transfer from the fluid air to the solid soil. In the past, ground velocity was measured using geophones. A modified off-the-shelf laser-Doppler vibrometer (LDV) is presently being used for ground velocity

measurements. The measurements are difficult because high atmospheric sound-pressure levels (80-90 dB) are necessary for good signal-to-noise in the acoustic-to-seismic coupling ratio measurements; the LDV also couples with the sound field. Good agreement has been reported among geophone and LDV measurements of vertical ground velocity for an impulsive atmospheric sound source and adequate agreement for a cw sound source in the frequency range 100-500 Hz. Use of the LDV for ground-geophone coupling measurements, also pertinent in seismology, will be discussed. [Work supported by ONR.]

2:30

GG7. Surface waves above porous ground surfaces. G. A. Daigle (Division of Physics, National Research Council, Ottawa, Ontario K1A 0R6, Canada)

In atmospheric acoustics, the subject of surface waves has been an area of discussion for many years. Theory for spherical wave propagation above an impedance plane describes the reflected field as a branch line integral (ground wave) with the possibility of a contribution from a pole. The contribution from the pole possesses the standard properties for a surface wave. Ground waves exist because curved wave fronts strike different parts of the surface at different angles of incidence and because the reflection coefficient is also a function of angle of incidence. Surface waves exist when the surface is sufficiently porous, relative to its acoustical resistance, that it can influence the airborne particle velocity near the surface and reduce the phase velocity of sound waves in air at the surface. This traps some of the sound energy in the air, regardless of the shape of the incident sound field, to remain near the surface as it propagates. Above porous grounds, the existence of surface waves has eluded direct experimental confirmation and indirect evidence for its existence has appeared contradictory. Experiments made here several years ago, but unreported, will be described and discussed in the light of some recent theoretical work. Further experiments are also planned.

2:45

GG8. The reflection of acoustic pulses from arrays of tubes. Heui-Seol Roh, James M. Sabatier, and Richard Raspet (Department of Physics and Astronomy and National Center for Physical Acoustics, University, MS 38677)

The theory of sound propagation in and the reflection of sound from rigid porous media is well established [K. Attenborough, *J. Sound Vib.* **99**, 521-544 (1985)]. This theory has been applied to regular arrays of straight tube to develop a low-reflectivity surface. Low reflectivities require high porosities in order to minimize the surface admittance. A 15.25-cm-i.d. tube terminated by arrays of rectangular tubes has been constructed to verify the theory in the limit of high porosity. Work is in progress to examine the effect of varying pore radius with depth.

Session HH. Psychological Acoustics V: Binaural Hearing and the Precedence Effect

Edward M. Burns, Chairman

*Department of Speech and Hearing Sciences, University of Washington, JG-15, Seattle, Washington 98195***Contributed Papers**

1:00

HH1. The effect of number of components and component spacing on the ability to detect a delayed component embedded in a diotic complex. M. A. Stellmack, R. H. Dye, and S. V. Jakubczak (Parmly Hearing Institute, Loyola University, 6525 N. Sheridan Road, Chicago, IL 60626)

The purpose of this study was to understand the processes that produce interference when observers attempt to lateralize a target component in the presence of other components. To this end, threshold interaural differences of time (Δ ITs) were measured for 753-Hz tones presented against a diotic background consisting of 2, 4, 6, or 8 additional components. The frequency spacing (Δf) of the components was 10, 25, 50, 100, or 200 Hz. A modified same-different task was employed in which a diotic 753-Hz tone was presented followed by either a diotic n -component complex or a complex containing a delayed (to the right channel) 753-Hz component. Stimuli were 200 ms in duration with 10-ms rise/decay times, with each component presented at 55 dB SPL. The largest thresholds were obtained when Δf 's were 25 or 50 Hz and the number of components in the background was large. Interestingly, thresholds fell as the frequency spacing was narrowed from 50 or 25 Hz to 10 Hz. In all cases, the obtained thresholds are larger than those obtained with a 753-Hz tone presented in isolation. The results are explained in terms of binaural processing that is spectrally synthetic for moderate to wide frequency separations and spectrally analytic at small frequency separations.

1:15

HH2. The effect of envelope power on the ability to lateralize three-tone complexes on the basis of interaural envelope delays. R. H. Dye, M. A. Stellmack, and A. J. Niemiec (Parmly Hearing Institute, Loyola University, 6525 N. Sheridan Road, Chicago, IL 60626)

In the present experiment, envelope power was varied by setting the starting phases of the three components of a complex 0° – 0° – 0° (high), 0° – 45° – 0° (medium), or 0° – 90° – 0° (low). Threshold interaural envelope delays were measured in a 2-AFC task as a function of frequency spacing (Δf = 25, 50, 100, and 200 Hz) with the carrier frequency (f_c) set to 4000 Hz. The level of each component was 50 dB SPL, and the signal duration was 200 ms with 10-ms linear rise/decay times. For comparison, thresholds were also measured for SAM 4000-Hz tones with reduced depths of modulation so that the effects of decreasing modulation depth and decreasing envelope power could be directly compared. Reducing envelope power had only a negligible effect on threshold interaural envelope delays for Δf 's of 200 and 100 Hz but a large and systematic effect at 50 Hz and especially at 25 Hz. For three of the five observers, the 0° – 90° – 0° , Δf = 25-Hz condition was so difficult that 75% correct could not be reached with delays as large as 2000 μ s. While reductions in the depth of modulation also elevated thresholds, the agreement was not particularly good between the data obtained with SAM tones and equal-amplitude components of the same Δf and comparable envelope power. In general, the results obtained here are consistent with binaural cross correlation of the outputs of envelope detectors.

1:30

HH3. Buildup and breakdown of the precedence effect. Richard L. Freyman, Rachel K. Clifton, Ruth Y. Litovsky, and Uma Balakrishnan (University of Massachusetts, Amherst, MA 01003)

Dynamic variations in the strength of echo suppression were investigated through earphone simulation of the free-field precedence effect. Each test stimulus consisted of two click pairs, with the interaural time parameters of the leading and lagging pairs reflecting stimuli originating from opposite sides of the head. Echo thresholds, the minimum lagging click delay required for subjects to report hearing an echo click, were obtained in two basic conditions: (a) test click presented in isolation, and (b) test click preceded by a train of clicks identical to the test click. The most striking finding was that the presence of the preceding click train usually elevated the echo threshold of the test stimulus relative to its threshold when presented in isolation, suggesting that echo suppression builds up during the click train. The magnitude of the effect was influenced by the number of clicks in the train and the period of silence between the end of the train and the test click. The effect is surprisingly persistent, lasting through several seconds of silence. Partial release from the built-up suppression occurred when the lead and lag locations of the test click were reversed from the clicks in the preceding train. [Work supported by NSF.]

1:45

HH4. Cross-frequency interactions in the precedence effect. R. K. Clifton (Psychology Department, University of Massachusetts, Amherst, MA 01003), P. M. Zurek, B. G. Shinn-Cunningham, and N. I. Durlach (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

The extent to which the precedence effect is observed when leading and lagging sounds occupy different spectral regions has been measured. Subjects, listening under headphones, were asked to match the lateral position of an acoustic pointer to that of a test stimulus composed of two binaural noise bursts with asynchronous onsets, parametrically varied frequency content, and different interaural delays. The precedence effect was measured by the degree to which the interaural delay of the matching pointer was independent of the interaural delay of the noise burst in the test stimulus. These results show an asymmetric frequency effect in which low frequencies dominate. For example, a stimulus composed of a high-frequency (1.25 kHz) leading burst and a low-frequency (0.45 kHz) lagging burst produces no precedence effect, whereas the reverse produces a strong precedence effect. These results are in qualitative agreement with those of Blauert and Divenyi [Acustica 66, 267–274 (1988)], despite numerous differences in stimuli and methods, including the restriction of this stimuli to the phase-sensitive ($f < 1.4$ kHz) region. [Work supported by NIH.]

HH5. Confirmation and rejection of the "precedence effect." Kourosh Saberi and David R. Perrott (Psychoacoustics Laboratory, California State University, Los Angeles, CA 90032)

A series of experiments designed to investigate the "precedence effect" in lateralization space were conducted using a two-alternative, forced-choice paradigm. Stimuli consisted of 40- μ s transients separated by inter-click intervals (ICIs) of 0.38–30 ms. In a two click paradigm (experiment 1) interaural differences of time (IDT) thresholds were determined independently at click positions 1 and 2. Thresholds measured at click 2 ranged from 12 μ s at extended (over 5 ms) and short duration (less than 1 ms) ICIs to a peak of 250 μ s at ICIs of 1.75–2.35 ms. Thresholds measured at click 1 were uniform across ICIs, constant at approximately 15–25 μ s. Experiment 2 consisted of a three click paradigm where the ICI between clicks 2 and 3 was held constant at a value where subjects in experiment 1 had peaked (1.75–2.35 ms), while the ICI between clicks 1 and 2 was varied from 0.38 to 10 ms. IDT thresholds were measured for click 3. Results show a U-shaped function for which thresholds drop below 30 μ s for click 3 at ICIs between 2–8 ms, with the effect returning at long and short duration ICIs. Experiment 3 consisted of click trains of $n = 2, 3, 4, 5, 6$, and 12 where ICI was held constant at subjects' peak value in experiment 1 and IDT thresholds were measured at the last click in the train. Results show a negatively decelerating function as n increases, with IDT thresholds falling from 250 down to 20 μ s. The most dramatic find in these series of experiments, however, was the surprising yet compelling dissipation of "suppression" in the temporal window of interest (1- to 5-ms ICIs) over multiple runs. Thresholds fell by 5 to 10 folds, matching values reported by Perrott *et al.* (in press) in the free field. Across variations of stimulus duration, form, intensity, or frequency, highly trained subjects showed no recovery of the precedence effect. [Work supported by NSF and NIH.]

HH6. Effects of roving level on binaural performance in normal-hearing and hearing-impaired listeners at 500 and 4000 Hz. P. Curliss, J. Koehnke, H. S. Colburn, and G. A. Owen (Boston University, 48 Cummington Street, Boston, MA 02215)

Measurements of NoS π thresholds and interaural time and intensity jnd's with and without a 10-dB roving level for both normal-hearing (4000 Hz) and hearing-impaired (500 and 4000 Hz) subjects have been obtained. This extends previous measurements of the effects of roving level on binaural detection and interaural discrimination at 500 Hz for normal-hearing listeners [J. Koehnke and H. S. Colburn, J. Acoust. Soc. Am. Suppl. 1 82, S108 (1987)]. All measurements were made using an adaptive, four-interval, two-alternative, forced-choice procedure and 1/3-oct noise bands. For detection, the target was a tone burst centered in the noise band. Binaural detection and interaural time discrimination show no effect of roving level at 500 Hz. However, interaural intensity jnd's at 500 Hz for the hearing impaired listeners are consistently larger with the roving level. In contrast to the data obtained at 500 Hz, at 4000-Hz performance of all the subjects for interaural intensity discrimination and NoS π detection is worse with the roving level. For interaural time discrimination, there is no apparent effect of the roving level at 4000 Hz. [Work supported by NIH.]

HH7. Detection and lateralization thresholds for repeated random waveforms in noise. Irwin Pollack (Mental Health Research Institute, University of Michigan, Ann Arbor, MI 48109-0720)

Interaural phase effects are reported for the lateralization and for the detection of repeated random waveforms in white noise. For lateralization, the primary component of interaural phase effects is the interaural phase of time-delayed signal and is relatively independent of the interaural phase of a nondelayed noise. In-phase signals yield substantially lower lateralization thresholds, at least for interaural delays up to 0.5–1.0 ms. This result is obtained with the following interaural noise relations: in phase, out of phase, random phase, and monaural noise. The importance of signal phase may relate to the search for moment-by-moment correlation by a binaural system with differential processing of waveform condensations and rarefactions. For detection, the primary component of interaural phase effects is related to the homophasic-antiphase combination of signal and noise. The magnitude of the interaural phase effects is substantially smaller for wideband repeated random waveforms than reported for pure tones. The smaller interaural phase effects may relate to the difficulty of establishing lateralization cues for wideband signals in the absence of a time delay.

WEDNESDAY AFTERNOON, 24 MAY 1989

NEWHOUSE II, ROOM 254, 1:00 TO 2:48 P.M.

Session II. Speech Communication VI: Perception and Analysis of Vowels

James E. Flege, Chairman

Department of Biocommunication, University of Alabama, University Avenue, Birmingham, Alabama 35294

Contributed Papers

III.1. The establishment of a new phonetic category by German learners of English. O. S. Bohn and J. E. Flege (Department of Biocommunication, University of Alabama, Birmingham, AL 35294)

This paper reports experiments examining the production of English vowels (/i, ɪ, e, æ/) and German vowels (/i/i, e, e:, a/) by two groups of

native German speakers differing in English-language experience. In experiment 1, German and English monolinguals read lists of English *bVt* words. Similar recordings were made of German front vowels by the German subjects. On the whole, duration and spectral measurements indicate that the distance between /i/ and /ɪ/ in an F_1/F_2 space is about the same in German, English, and native Germans' English. The vowels /i/ and /ɪ/ of English and German differed acoustically. Both experienced and inex-

perienced Germans approximated but did not reach the English monolinguals' values for /i/ and /ɪ/. Experienced but not inexperienced native Germans differentiated English /æ/ and /ɛ/ acoustically, probably because German lacks an /æ/ category. The acoustic results of experiment 1 were consistent with experiment 2, which examined monolingual English listeners' categorization of the German speakers' English *bVt* words. The results support Flege's [J. Phonetics 15, 47-65 (1987)] model of L2 learning which predicts that extended foreign language experience will lead to the establishment of phonetic categories for new (e.g., /æ/) sounds in L2. [Work supported by NIH Grant 20963.]

1:12

II2. The perception of English vowels by native speakers of Spanish. J. E. Flege and O. -S. Bohn (Department of Biocommunication, University of Alabama, University Station, Birmingham, AL 35294)

This study examined native Spanish speakers' perception of four English vowels (/i, ɪ, ɛ, æ/). In experiment 1, subjects used letters ((i,e,a,o,u)) to identify the vowels in *beat*, *bit*, *bet*, *bat* or responded "none" if they did not hear a Spanish vowel. The pattern of responses was unsurprising: mostly (i) for the English vowels /i/ and /ɪ/, (e) for the English /ɛ/s, and (a) for /æ/. Subjects who could speak English responded "none" significantly more often than Spanish monolinguals for all four words, suggesting they had begun to differentiate the English vowels from their nearest phonemic counterpart in Spanish. In experiment 2, subjects identified the members of continua which varied spectral quality (11 steps) and vowel duration (3 steps). Like native speakers, most Spanish subjects showed clear crossovers when identifying stimuli ranging from /bet/ to /bæt/, probably because the continuum endpoints were identified with different Spanish vowels (viz., /e/ and /a/). Only six (30%) of the Spanish subjects, however, showed clear crossovers for a /bit/-to-/bit/ continuum, probably because the endpoints were identified with reference to a single Spanish vowel (/i/). The pattern of identification responses did not change systematically when stimuli differing in spectral quality were blocked on vowel duration in experiment 3. The results were interpreted to mean that even experienced Spanish speakers of English may not establish phonetic categories for "new" English vowels such as /ɪ/ and /æ/. [Work supported by NIH Grant 20963.]

1:24

II3. S'pose a vowel is lost. Joann Fokes (School of Hearing and Speech Sciences, Ohio University, Athens, OH 45701) and Z. S. Bond (Department of Linguistics, Ohio University, Athens, OH 45701)

Various types of reduction can be observed in casual speech, one being vowel loss in unstressed syllables as in "s' pose." The question, how is the phonetic detail of this specific reduction implemented, is the focus of the study. Three types of test words were selected for investigation, initial /sp/ and /sk/ clusters, two and three syllable words beginning with an unstressed /a/ + V followed by /p/ or /k/ in the stressed syllable, and a word with a cluster created by the loss of the vowel, such as *sport*, *support*, and *s'port*. Male subjects read each test word in context five times. Measurements included (1) duration of the /s/, (2) duration of the stop closure, (3) voice onset times, and (4) total time from the beginning of the /s/ to the onset of the stressed vowel. Results indicated two different speaker patterns but consistency within each subject. Differences between the subjects occurred in the length of the /s/ in all types of test words, voice onset time, and the total time. For one of the speakers, voice onset time was relatively short regardless of the cluster or word types. For another speaker, VOT was short for the /sp/ clusters while values for the /sk/ cluster, for the created clusters, and for the two and three syllable words became increasingly longer. This speaker differentiated between real clusters and created clusters by means of voice timing.

1:36

II4. The effects of relative formant intensity on vowel perception revisited. Kam-Cheong Tsoi (Department of Psychology, SUNY at Buffalo, Amherst, NY 14260)

The effects of relative formant intensity on the phonetic identity of vowels were investigated in the present study. Ambiguous stimuli which were near the phonetic category boundary of /a/-/ɔ/, /u/-/ʊ/, and /ɛ/-/æ/ series were used to construct experimental stimuli by varying the intensity ratio between *F*1 and *F*2. The results showed that the perceived identity of these three sets of stimuli shifted towards /a/, /ʊ/, and /æ/ when the intensity of *F*2 was increased, and shifted towards /ɔ/, /u/, and /ɛ/ when the intensity of *F*2 was decreased. The magnitude of effect was largest in stimuli of the /a/-/ɔ/ series, and smallest in stimuli of the /ɛ/-/æ/ series. The results suggest that (1) the relative intensity of formants does affect the phonetic identity of the stimuli, and (2) the frequency range within which spectral integration between adjacent formants can take place is not limited to 3 Barks, but follows a continuous probability function. [Work supported by NIH.]

1:48

II5. Tongue positioning and dynamics during production of tense and lax vowels in CVC contexts. P. A. Dagenais (Department of Biocommunication, University of Alabama, P. O. Box 503, UAB Station, Birmingham, AL 35294)

A comparison between tense and lax vowel pairs was undertaken using simultaneous recordings of tongue movements via glossometry and audio signals. A normal speaker recorded five repetitions each of three pairs of tense-lax vowels (i-ɪ, æ-ɛ, u-ʊ) in plosive context (ɔpvp, ɔtv, ɔkvk). Measures for maximum tongue displacement, durations and velocities of physiological onglide, and target and offglide movements relative to the acoustic release of the initial plosive were taken. Greater maximal displacement from a central position was noted for all tense vowels. In pvp contexts, consistent preplosive anticipatory positioning was found for all vowels, and maximal tongue positioning for the lax vowels was achieved 50 ms prior to the tense vowels. For both lingual plosive contexts, tense vowel movements had longer total durations but slower onglide and offglide velocities than their lax counterparts. These results will be discussed in terms of biomechanical restrictions for production and the effects of perceptual saliency and coarticulation. [Work supported by NIH Grant NS24697.]

2:00

II6. Relationship between vowel height and speaking rate in American English. H. S. Gopal, Joyce Manzella, and Lillian Stuman (Department of Speech and Hearing Sciences, University of California at Santa Barbara, Santa Barbara, CA 93106)

The phonetic dimension of vowel height seems to be represented reliably and robustly by the difference between Bark-transformed *F*1 and *F*0 [A. K. Syrdal and H. S. Gopal, J. Acoust. Soc. Am. 79, 1086-1100 (1986)]. This study examined the relationship between speaking rate and two acoustic-auditory correlates of vowel height. Measurements were made of *F*0, *F*1, vowel duration, and sentence durations of /pVC/ target syllables containing one of four pairs of high-low vowels and voiced or voiceless stops or fricatives. Target syllables were produced in sentence context at three different speaking rates by two male and two female native American English speakers. Results will be reported and discussed for two correlates of vowel height: changes in *F*1 and changes in the difference between Bark-transformed *F*1 and *F*0 as a function of speaking rate.

II7. Minimalist synthesis of /bVb/ stimuli. Jean E. Andruski and Terrance M. Nearey (Department of Linguistics, University of Alberta, Edmonton, Alberta T6G 2E7, Canada)

Vowels both in isolation and in /bVb/ syllables can be well identified from brief (30 ms) windowed portions taken from the beginning (head) and end (tail) of the syllable [T. Nearey and J. Andruski, J. Acoust. Soc. Am. Suppl. 1 83, S83 (1988)]. Despite similar confusion matrices, formant measurements of windowed /bVb/ data show substantial undershoot compared to initial and final targets of isolated vowels. It is of interest to construct stimuli that are minimally specified but are nonetheless categorized like more complex natural stimuli. Vowels were synthesized with linear transitions based on fundamental and formant frequency measurements from four brief sections of the windowed /bVb/ stimuli. (These sections correspond roughly to transition onset, initial vowel target, final vowel target, and transition offset.) Preliminary listening tests indicate that these simple stimuli are well identified. Results from experiments comparing listeners' error patterns for such minimalist synthetic syllables with those for windowed naturally produced syllables will be presented.

II8. Modeling of contextual effect based on spectral peak interaction. Masato Akagi (ATR Auditory and Visual Perception Research Laboratories, Inuidani, Seikacho, Kyoto, 619-02 Japan)

This paper presents a model of a lower level contextual effect that can cope with coarticulation problems, especially vowel neutralization. The model is constructed to overshoot spectral peak trajectories based on spectral peak interaction, assuming that the lower level contextual effect is represented as the sum of interaction between each spectral peak pair. The interaction function is determined experimentally in order to reduce

the distance between a real spectral peak and its target which is a spectral peak mean computed for vowel uttered in isolation. The interaction function thus determined suggests that: (1) there can be a time-frequency lateral inhibition in the auditory system like that on the retina in the visual system, (2) the interaction function is consistent with the results of psychoacoustic experiments concerning the assimilation and/or contrast effect using paired single formant stimuli, and (3) the contextual effect between adjacent phonemes can be represented as the sum of the assimilation and/or contrast effects between each spectral peak pair. Applying the determined interaction function to real speech data to cope with coarticulation problem, spectral peak trajectories overshoot, spectral peaks at the vowel center approach their own targets, and the distance between each vowel category pair increases.

II9. Experimental evidence for the unit of bird song production. Jeffrey Cynx (Rockefeller University Field Center, Millbrook, NY 12545)

An experimental procedure was developed to determine the real-time nature of birdsong production. In a given trial, a burst of strobed light, directed at a male zebra finch (*Poephila guttata*), was triggered at a random point in his courtship song. Repeated trials allowed the formulation of three rules governing song production: (1) A zebra finch can stop song production in the midst of song, (2) the motor unit for song production is quantal, and (3) the quantal unit of song production is largely identical to a syllable, as read off a sound spectrogram. The comparative generality of the rules was tested by replicating results with another two other oscine species, the song sparrow (*Melospiza melodia*) and the canary (*Serinus canarius*). The results place constraints on neural models of song production, and suggest invariance between the units of song perception and production. [Research supported by NIH and the Cary Charitable Trust.]

WEDNESDAY AFTERNOON, 24 MAY 1989

REGENCY C, 1:00 TO 3:00 P.M.

Session JJ. Underwater Acoustics V: Acoustic Bottom/Interface Properties

Ronald Dicus, Chairman
Naval Research Laboratory, Washington, DC 20375

Chairman's Introduction—1:00

Invited Paper

1:05

JJ1. Geoacoustic properties of the seabed sediment critical to acoustic reverberation at 50 to 500 Hz: A preliminary data set. Tokuo Yamamoto, Morris Schulkin (Geo-Acoustics Laboratory, Applied Marine Physics Division, University of Miami, Miami, FL 33149-1098), and Richard Bennett (NORDA Code 360, Stennis Space Center, MS 39529)

As much as 80% of acoustic reverberation is produced through bottom interactions. Deep-sea sediment have high porosity (or low velocities), therefore, a waveguide is often formed from strong bottom interactions. In addition to the bottom roughness, the random variation in the geoacoustic properties (the compressional wave velocity in particular) of the sediment is responsible for generation of the incoherent propagation/acoustic reverberation. Our existing data [e.g., Bennett *et al.*, *Handbook of Geophysical Expro.* (1983)] indicate that very strong spatial variations (vertical and horizontal) usually exist in the deep-sea sediments. A

preliminary analysis of these geological data by the Biot-Yamamoto theory [Yamamoto and Turgut, J. Acoust. Soc. Am. **83**, 1744-1751 (1988)] reveals that strong velocity variations, as much as 20% in velocity magnitude, usually exist in the deep-sea muds at horizontal and vertical scales of 0.3 to 30 m, which is very critical to acoustic reverberation at 50 to 500 Hz. [Work supported by ONR Code 11250 A.]

Contributed Papers

1:30

JJ2. High-resolution investigations of seismoacoustic propagation in shallow-water areas. George H. Sutton, Noël Barstow, Jerry A. Carter (Rondout Associates, Incorporated, P. O. Box 224, Stone Ridge, NY 12484), John I. Ewing (Woods Hole Oceanographic Institution, Woods Hole, MA 02543), and David Harris (Hawaii Institute of Geophysics, University of Hawaii, 2525 Correa Road, Honolulu, HI 96822)

Data from a newly developed 30-element accelerometer/hydrophone array are used to study seismoacoustic propagation in shallow-water locations off New Jersey and Martha's Vineyard. The elements of the array have 1-m spacing; each consisting of three orthogonal accelerometers (flat from 2 to 500 Hz), a hydrophone, and a vertical direction sensor. The 120 accelerometer/hydrophone signals and 30 vertical direction signals are digitized and transmitted via 1 km of fiber optic cable to a PC-type recording system. The A/D converters digitize all hydrophones at 2048 samples/s and all accelerometers at 512 samples/s. Good coupling to the bottom is obtained since sensor symmetry has been maximized; coupling to the water has been minimized; and density is matched to the sediments. There are 10-m extensions between the 30-m sensor section and the A/D converter housing at either end of the array. Units containing port-starboard paired, shotgun-shell sources were mounted within the array in these extensions and fired through the array circuitry, providing precise time and distance control and reverse profile capability. Differenced shot pairs emphasize transverse horizontal shear (*SH*) and Love wave signals; summed pairs emphasize *P*, *SV*, and Rayleigh/Stoneley/Scholte waves. Shear boundary wave energy is observed with frequencies greater than 60 Hz and velocities and wavelengths as low as 20 m/s and 1 m, respectively. Considerable energy appears to result from lateral heterogeneity and/or anisotropy. [Research supported by ONR.]

1:45

JJ3. Low-frequency measurements of bottom reflectivity at a thin sediment site in the North Pacific. Patricia L. Gruber (AT&T Bell Laboratories, 1201 S. Hayes Street, Arlington, VA 22202) and Ronald L. Dicus (Naval Research Laboratory, Washington, DC 20375)

Bottom reflectivity measurements were made in the North Pacific, during Pacific Echo I, using a low-frequency, towed source. A MK VI source transmitted 50-s FM sweep signals (5 to 15 Hz) to a vertical array of 16 hydrophones. The FM ramps were deconvolved and beamformed to separate direct and bottom arrivals in time and angle. In addition, ray theory predictions of the arrival structure, scaled by spreading loss, were beamformed. A comparison between the time integrated energy in the measured and theoretical bottom arrivals, led to a measure of the bottom reflectivity. FM ramps were processed along the source ship track to vary the bottom grazing angle from 10 to 75 deg. Reflectivity as a function of bottom grazing angle, smoothed over angle to reduce variance, will be presented. Theoretical computations of reflectivity from a simulated bottom that includes a sediment layer with sound-speed gradient and an elastic basement, with shear and compressional speed gradients and attenuations, will be presented. Theory and measurements are compared to infer the sediment thickness and the wave speed profiles in the upper 200 m of the oceanic crust.

2:00

JJ4. Bottom shear modulus profiler (BSMP) measurements of the shallow-water seabed geoacoustics properties in Japan. Mohsen Badiy (University of Miami, Miami, FL 33149 and Nakajima Institute of Geophysics, Nakajima City, Ishikawa Prefecture, Japan), Tokuo Yamamoto (University of Miami, Geoacoustics Laboratory, 4600 Rickenbacker Causeway, Miami FL 33149-1098), Hideo Suzuki (Ono Sokki Co. Ltd., Tokyo 163, Japan), and Hiroshi Kanai (Tohoku University, Sendai, Japan)

The method of bottom shear modulus profiler (BSMP) was used to measure the seabed properties of several locations in Japan. In this method, the seabed admittance, which is defined as the ratio of the wave-induced particle motion to the wave pressure on the seafloor is measured [T. Yamamoto and T. Torii, Geophys. J. K. Astron. Soc. **85**, 413-431 (1986)]. These data are then used as input to a linear inversion scheme for calculation of the shear modulus profile. The collected data are presented in a complete form of shear and compressional wave velocity profiles. An attempt has been made to improve the numerical analysis of the inverse method used in the BSMP software. By applying a tapering Hamming window in the truncation of the order employed in the singular value decomposition (SVD) technique, the truncated expansion equation was modified and the higher-order eigenvalues were used stably. A numerical test as well as the analysis of the data using this method is presented.

2:15

JJ5. Results from two shallow-water experiments utilizing synthetic bandwidth ("stepped FM") signals to estimate compressional wave speed in the seabottom. Neil J. Williams and Harry A. DeFerrari (Department of Applied Marine Physics, RSMAS, University of Miami, Miami, FL 33149)

Two experiments were performed to investigate the utility of synthetic bandwidth or "stepped FM" signals [H. A. DeFerrari and H. B. Nguyen, J. Acoust. Soc. Am. Suppl. **1** **82**, S74 (1987)] for acoustic remote sensing of geoacoustic properties of the seabottom. In each experiment, pseudo-random codes with carrier frequencies ranging from 100 to 1600 Hz were transmitted consecutively at a fixed range to a moored vertical array. After every other complete sweep of the entire bandwidth, the range was increased and the transmission was repeated. The experimental sites were 42 nautical miles SE of Corpus Christi, TX and 75 nautical miles E of Lewes, DE. The water depth in each case was about 60 m and the range from source to receiver varied from 300 m to 2 km. The data have been reduced to pulse responses with a resolution of 6 ms. This, combined with the angular resolution of the array, aids in the identification of eigenrays and inversion of the data to yield estimates of compressional wave speed in the seabottom. [Work supported by ONR.]

2:30

JJ6. Gaussian beam algorithm for bottom reflection and head waves due to extended aperture sources. X. J. Gao and L. B. Felsen (Department of Electrical Engineering/Computer Science, Weber Research Institute, Polytechnic University, Farmingdale, NY 11735)

Gaussian beams furnish a useful algorithm for high-frequency sound propagation in complex environments but in their paraxial implementation, they are flawed by spectral defects that obscure certain wave phenomena. Critical reflection and head wave generation due to a fast bottom belong in this category. In a separate study, methods for "fleshing out" the deficient spectrum for the prototype structure of a homogeneous ocean and a homogeneous fluid bottom, have been explored with excitation due to a line source [X. J. Gao *et al.*, 2nd IMACS Symposium, Princeton University (March 1989)]. In the presentation here, these methods of excitation from a continuously distributed extended aperture source and from an array of line sources have been applied. In the Gaussian beam modeling, not only are the *ad hoc* discretization that involves arbitrary parameters employed but the self-consistent Gabor phase space stacking are also employed. It is found that distributed sources deemphasize the above-noted spectral defects but do not remove them entirely. [Work supported by ONR.]

2:45

JJ7. Propagation of Rayleigh and Scholte waves along edge of quarter-space. Jacques R. Chamuel (Sonoquest Advanced Ultrasonics Research,

P. O. Box 153, Wellesley Hills, MA 02181-5339) and Gary H. Brooke (Defence Research Establishment Pacific, FMO Victoria, British Columbia VOS 1B0, Canada)

A surprisingly large leaky Rayleigh wave component has been observed propagating along the edge of a laboratory nonsaline-ice *quarter-space* in water with both source and receiver positioned along the edge of one of the two surfaces of the quarter-space. Studies on different solids (aluminum, Plexiglas, limestone) indicate that all surface and interface edge waves travel slower than their corresponding waves on a solid half-space. The measured ratio of the edge Rayleigh wave velocity to the half-space Rayleigh wave velocity is 0.9617 for aluminum 6061, 0.9665 for Plexiglas, and 0.9954 for limestone. Ultrasonic experimental results are presented on edge Rayleigh waves, and on edge leaky Rayleigh waves and Scholte waves for the liquid/solid case. The edge wave has no geometrical spreading and dominates the received signal when the receiver is moved slightly away from the water loaded edge of the quarter space. The Rayleigh waves along the horizontal and vertical faces at the edge are out of phase influencing the radiation of the leaky Rayleigh wave for the water-loaded case. Near the edge it is observed that a decrease in the Scholte wave signal is accompanied by an increase in the leaky Rayleigh wave signal. The leaky Rayleigh wave velocity for the water/ice is slightly smaller than the shear wave velocity at the onset of the signal. [Work supported by DREP and ONR.]

WEDNESDAY AFTERNOON, 24 MAY 1989

COMSTOCK B, 1:45 TO 3:00 P.M.

Session KK. Bioresponse to Vibration III and Physical Acoustics VI: Biomedical Acoustics

Wesley L. Nyborg, Chairman

Physics Department, University of Vermont, Burlington, Vermont 05405

Contributed Papers

1:45

KK1. A comparison between low-frequency sound absorption in seawater and high-frequency sound absorption in serum albumen. David G. Browning (Code 3131, Naval Underwater Systems Center, New London, CT 06320) and Robert H. Mellen (Kildare Corporation, 95 Trumbull Street, New London, CT 06320)

Although large scale (> 1000 km), low-frequency ($< 20\,000$ Hz) sound absorption measurements in seawater [Mellen *et al.*, *J. Acoust. Soc. Am.* **74**, 987-993 (1983)] might appear to have nothing in common with high-frequency sound absorption in serum albumen [Barnes *et al.*, *J. Acoust. Soc. Am.* **83**, 2393-2404 (1988)], the results show some intriguing similarities that may be due to a common chemistry. Specifically, both show a pH dependence, a buffered reaction involving a carboxyl group, and the need of calcium to promote the principal reaction. However, boric acid, which is essential for seawater absorption, is apparently not a factor in serum absorption. A comparison is presented in the hope of stimulating a dialogue with the bioacoustics community to explore a possible linkage.

2:00

KK2. The influence of biophysical conditions on hemolysis near ultrasonically activated gas-filled micropores. Douglas L. Miller and Ronald M. Thomas (Battelle Pacific Northwest Laboratories, P.O. Box 999, Richland, WA 99352)

Hemolysis in 0.5% suspensions of canine erythrocytes exposed to 1.9-MHz ultrasound in the presence of 3.7- μ m-diam gas bodies in filter micropores (Nucleopore[®]) was investigated as a function of spatial peak intensity for varied biophysical conditions. Significant hemolysis was observed

above 90 mW/cm², which increased to 50% at 500 mW/cm², for 30-min exposure at 37 °C in isotonic phosphate buffered saline. Hemolysis increased with exposure duration (4-64 min) but did not greatly change with exposure temperature (25, 37, 48 °C), or prior heat treatment (48 °C for 30 min). The temperature results were especially interesting because increased temperatures might have been expected to increase the sensitivity of the cells to the shear stress generated in acoustic microstreaming flow near the ultrasonically activated gas bodies. Increasing the viscosity (1, 2, 4 cP) of the medium decreased the effect, while variations in osmolarity (180, 290, or 580 mOsm) or density (1.03 or 1.12 g/cm³) had little influence on the results. Increasing the density of the medium might have been expected to reduce the effectiveness of the exposures, since the radiation force, which theoretically gathers cells to the gas bodies, vanishes for isopycnic conditions. [Work supported by PHS Grant CA42947 awarded by the National Institutes of Health.]

2:15

KK3. Measurement of *in vivo* lung acoustic impedance for animals using random noise and the two microphone technique. J. E. Sneckenberger and Cheryl Chandler (Department of Mechanical and Aerospace Engineering, West Virginia University, Morgantown, WV 26506)

The *in vivo* acoustic impedance of rats' lungs was measured using random noise and the two microphone transfer function method for calculating impedance. This method eliminates the need for the sensitive pressure transducers used in the conventional measurement of respiratory impedance. Measurements were performed on control rats and rats that had, 5 months prior to the experiments, been exposed to coal dust for 2 weeks. Random noise, flat to 6.4 kHz, was propagated down a three section impedance tube. The experiments began with the insertion of a tra-

cheal tube through the rat's mouth to a depth of approximately 3/4 in. into the trachea. Sound traveling through the impedance tube and tracheal tube and reflecting from the lung was measured with the two microphones and recorded by a two channel FFT analyzer. The analyzer calculated the transfer function between the microphones and, from this transfer function, the complex lung acoustic impedance was calculated. The impulse response was also computed from the transfer function, and was further processed, by the Ware-Aki area inversion algorithm, to produce the effective airway cross-sectional area as a function of depth into the lung from the trachea.

2:30

KK4. Target strengths of Antarctic krill. K. G. Foote (Institute of Marine Research, 5024 Bergen, Norway), I. Everson, D. G. Bone, and J. L. Watkins (British Antarctic Survey, High Cross, Madingley Road, Cambridge CB3 0ET, United Kingdom)

Encaged aggregations of *Euphausia superba* have been ensonified at 38 and 120 kHz and the echo energy measured. Derived estimates of target strength are considerably lower than previously measured values.

2:45

KK5. Study of intensity in the focal lobe for a focused ultrasonic stone removal device. V. R. Singh and Ravinder Agarwal (Instrumentation, National Physical Laboratory, New Delhi-110012, India)

In modern surgical practice, the operative procedures for stone removal are still finding difficulty in their acceptance by the patients. Thus it gives motivation for finding a new and alternative, noninvasive technique for the removal of kidney stones. Focused ultrasound is used in the disruption of such stones. Intensity pattern parameters in the focal lobe are studied for a transducer used for removal of kidney stones.

WEDNESDAY AFTERNOON, 24 MAY 1989

CROUSE AUDITORIUM, 3:30 P.M.

Plenary Session

W. Dixon Ward, Chairman
President, Acoustical Society of America

Presentation of Awards

R. Bruce Lindsay Award to Mark F. Hamilton
Gold Medal to Lothar W. Cremer

Musical Demonstration and Lecture

Professor William Headlee will give a recital on the Syracuse University Holtkamp Organ, preceded by a short lecture on its history.

WEDNESDAY PM

Session LL. Engineering Acoustics VI: Transducers and Arrays

Sung H. Ko, Cochairman

Naval Underwater Systems Center, New London, Connecticut 06320

Howard H. Schloemer, Cochairman

*Naval Underwater Systems Center, New London, Connecticut 06320***Contributed Papers****8:00****LL1. History of the flextensional transducer.** Kenneth D. Rolt^(*)
(Department of Ocean Engineering, MIT, Cambridge, MA 02139)

Although the invention of the flextensional acoustic transducer is usually attributed to William J. Toulis, it was actually invented at the U. S. Naval Research Laboratories, Washington DC, in 1929 by Harvey C. Hayes. It was intended as a foghorn for the Lighthouse Service. This paper will review Hayes' flextensional from invention to disappearance (c. 1936), and discuss its reinvention and development by Toulis and Frank R. Abbott in the 1950s. ^(*) Presently on educational leave from, and work supported by Acoustic Engineering, Raytheon Company, Submarine Signal Division, Portsmouth, RI 02871-1087.

8:15**LL2. A theoretical model for an array of flextensional transducers.** Richard J. Brind (Admiralty Research Establishment, Portland, Dorset DT5 2JS, England)

Acoustic interaction effects can seriously degrade the performance of arrays of sonar transducers if the efficiency of the elements is high and the interelement spacing is small. An experimental array of four 300-Hz flextensional transducers was deployed in Loch Goil in Scotland to investigate these effects. Here, an equivalent circuit model for the transducers is developed and a theory for the interaction effects presented. The model is validated against the array measurements taken in Loch Goil. The theory was used to predict admittance loops and farfield beam patterns for unsteered and steered arrays with half-wavelength and quarter-wavelength element spacings. An interactive program has been written to perform these calculations on an HP45 computer. Farfield beam patterns were obtained directly with a hydrophone lowered from the surface, and were calculated from nearfield measurements. The agreement between the predictions and the measurements was very good. The interaction effects in an array with half-wavelength spacing were small. With quarter-wavelength spacing, interaction effects were severe, giving rise to a 2.5-dB reduction in transmitting voltage response and narrow frequency bands where the conductance of one of the elements was dangerously low.

8:30**LL3. Ultra-high-power superconducting transducers.** Osman K. Mawardi (Collaborative Planners, Inc., Cleveland, OH 44106)

Recent developments in high-temperature superconducting materials make it practical to investigate the feasibility of very high-power superconducting acoustic transducers capable of generating several hundreds of acoustic kilowatts. A proposed scheme for such a transducer that consists of a variation of a linear motor is described in this paper. Borrowing from the highly developed technology of linear electrical machinery, it is

shown that a very compact transducer can be constructed with an energy density one order of magnitude higher than that of a magnetostrictive device. A detailed design and analysis has been completed for a transducer with a rating of 250 kW and an operating frequency of 50 kHz.

8:45**LL4. Analysis of magnetostrictive transducers by the ATILA finite element code.** F. Claeysen, D. Boucher (GERDSM, DCAN Toulon, DCN, Le Brusc, 83 140 Six-Fours-Les-Plages, France), K. Anifrani (SINAPTEC, 121 Rue Chanzy, 59620 Hellemmes, France), R. Bossut, and J. N. Decarpigny (ISEN, 41 Boulevard Vauban, 59 046 Lille Cedex, France)

New magnetostrictive rare-earth-iron alloys offer an attractive opportunity for the design of high-power low-frequency transducers [F. Claeysen, J. Acoust. Soc. Am. Suppl. 1 **81**, S89 (1987)]. In order to optimize the design of this particular class of transducers, a model based on a new variational principle has been derived following a classical finite element method approach. This model describes the three-dimensional dynamic behavior of heterogeneous electrochemically coupled structures. Due to the reduced-scalar-potential formulation of the magnetic field, the model has been relatively easily implemented within the ATILA finite element code. In this paper, a general view of the method is given and first computation results on test structures are presented.

9:00**LL5. Improvement of a magnetostrictive length-expander transducer by use of a grain-oriented material.** F. Claeysen, D. Boucher, and S. Faure (GERDSM, DCAN Toulon, DCN, Le Brusc, 83 140 Six-Fours-Les-Plages, France)

In order to prove the ability of new magnetostrictive rare-earth-iron alloys for high-power low-frequency transmitting transducers, a first prototype of a length-expander projector had been realized in 1987. Although the active material was a random-oriented alloy (R), high-power measurement results were quite satisfactory [F. Claeysen, J. Acoust. Soc. Am. Suppl. 1 **81**, S89 (1987)]. On the other hand, an active material characterization work [F. Claeysen, J. Acoust. Soc. Am. Suppl. 1 **83**, S20 (1988)] has led to theoretical predictions showing the advantages of grain-oriented alloys (G.O.) compared to the random-oriented alloy. In addition to the increase of acoustical power, a lower resonance frequency and a better effective coupling factor are expected. In this paper, the theoretical improvement of a transducer due to the use of G.O. alloy is checked with the in-air and in-water measurements of the prototype transducer mounted with a G.O. material.

LL6. Finite element modeling of piezoelectric transducers including internal losses. B. Dubus, J. C. Debus, J. N. Decarpigny (Institut Supérieur d'Electronique du Nord, 41 Boulevard Vauban, 59046 Lille Cedex, France), and D. Boucher (Groupe d'Etude et de Recherche de Détection Sous-Marine, Le Brus, 83140 Six Fours les plages, France)

An actual radiating sonar transducer is always generating heat, due to internal losses and to the friction between the different parts of the structure. Thus its efficiency is lower than 1 and a temperature rise can occur. To describe these phenomena, the finite element code ATILA [Decarpigny *et al.*, J. Acoust. Soc. Am. **78**, 1499 (1985)] has been improved by including the effects of internal losses, using a complex Young's modulus for isotropic materials and complex tensors for piezoelectric ceramics. Then, electrical and thermal measurements have been performed on various sandwich transducers, at different drive levels, to determine the ceramics loss angles. Finally, the electrical impedance, transmitting voltage response, and mechanical displacement field have been computed for a multifrequency Tonpilz transducer [Boucher *et al.*, J. Acoust. Soc. Am. Suppl. **1** **80**, S27 (1986)] and a ring transducer [Bossut *et al.*, J. Acoust. Soc. Am. Suppl. **1** **74**, S23 (1983)] radiating in water, and the theoretical results have been compared to experimental values, showing a reasonable agreement which will be carefully discussed.

9:30

LL7. A Poisson-coupled plastic projector. G. W. McMahon and J. B. Franklin (Defence Research Establishment Atlantic, Box 1012, Dartmouth, Nova Scotia B2Y 3Z7, Canada)

An underwater electroacoustic projector is described in which a hollow tube of plastic is driven axially in order to excite its radial resonance by Poisson coupling. The driver is a piezoceramic stack that connects to the plastic tube via stiff metal end plates. The projector was designed using the finite-element program MAVART to predict performance and enable a suitable selection of dimensions for an experimental model. The projector is 18 cm long, 11.4 cm in diameter, and weighs 0.7 kg. Although no well-defined resonance is exhibited, efficient operation is predicted from 2200 to 3300 Hz, with a source level up to 191 dB re:1 μ Pa @ 1 m. The measured performance of the experimental unit is compared with MAVART predictions.

9:45

LL8. Finite element modeling and array interaction modeling of the ARE low-frequency flextensional transducer. Richard J. Brind (Admiralty Research Establishment, Portland, Dorset DT5 2JS, England)

A requirement for underwater acoustic sources with higher source level and wider bandwidth at lower frequencies has, in the past 5 years, lead to interest in flextensional transducers. A low-frequency prototype has been developed, manufactured, and tested at ARE(Portland); the technology is the subject of a license agreement with industry. In this paper, finite element (FE) analyses performed in support of the prototype development using PAFEC software are described. PAFEC is a commercially available FE package with the facility to model piezoelectric materials, and exterior fluid loading by the doubly asymptotic approximation. In-air and in-water resonant frequencies are well predicted, as is the effect of hydrostatic pressure on stack prestress. The transducer is subjected to large stresses during assembly, and predictions for these have also been obtained. Problems remaining in the FE modeling, and improvements in the capabilities of PAFEC under development to overcome them, are discussed. An equivalent circuit model for an array of these flextensional transducers is described, and a comparison of admittances and farfield predictions with measurements presented.

10:00

LL9. Finite element modeling of the ARE low-frequency flextensional transducer. Richard J. Brind (Admiralty Research Establishment, Portland, Dorset DT5 2JS, England)

A requirement for underwater acoustic sources with higher source level and wider bandwidth at lower frequencies has, in the past 5 years, lead to interest in flextensional transducers. A low-frequency prototype, which consists of a GRP shell and convention ceramic stacks, has been developed, manufactured, and tested at ARE(Portland); the technology is the subject of a license agreement with industry. In this paper, finite element (FE) analyses performed in support of the prototype development using PAFEC software are described. PAFEC is a commercially available FE package with the facility to model piezoelectric materials, and exterior fluid loading by the doubly asymptotic approximation. In-air and in-water resonant frequencies are well predicted, as is the effect of hydrostatic pressure on stack prestress. The shell is subjected to large interlaminar shear stresses during stack insertion, and predictions for these have also been obtained. Work is in hand at PAFEC to add the capability to couple a Helmholtz integral equation representation for the exterior fluid to the existing finite element package for the structure. This will enable source level and bandwidth calculations to be made, which, at present, cannot be accurately predicted.

10:15

LL10. Minimax shading of arbitrary arrays. Thomas E. Bordley (Naval Research Laboratory, Code 5123, Washington, DC 20375-5000)

An iterative algorithm for minimizing the maximum sidelobe level of an arbitrarily shaped array over bands of frequencies and directions is presented. The approach is novel both in providing an optimal Tchebyshev solution for nonuniformly spaced, multidimensional arrays and in providing a solution over frequency bands. At each iteration, the algorithm generates a descent direction minimizing the response at the highest sidelobes. The method is applicable both to plane-wave beamforming and to matched field processing. In this presentation, the trade-offs between beamwidth, bandwidth, and sidelobe level are considered for the conventional beamformer for a variety of nonuniformly spaced arrays.

10:30

LL11. Random linear projector arrays with reduced acoustic interaction. P. M. Joseph, P. R. Saseendran Pillai, and C. S. Sridhar (Department of Electronics, Cochin University of Science and Technology, Cochin 682 022, India)

Acoustic interaction effects among the elements of closely packed sonar projector arrays should be minimized considerably, as it reduces the acoustic power output and deteriorates the beam characteristics, at higher drive levels. This effect can be minimized by restructuring the array format with appropriately chosen interelement spacing [P. M. Joseph and P. R. Saseendran Pillai, in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing* (IEEE, New York, 1988), pp. 2697-2700] and further reduction in interaction effect can be achieved by incorporating the nonuniform array concept. The appropriate interelement spacing for this nonuniform array can be determined by using total enumeration method. But this method cannot be recommended for arrays comprised of large number of elements, as it is a time consuming and tedious process. In the approach presented in this paper, the suitable interelement spacings that reduce the interaction are determined by using Gaussian distributed random numbers with the element spacing of the restructured array as mean and by retaining the array dimension same as that of the restructured array. Beam pattern and array gain are found to be little affected in the randomly distributed element configuration.

LL12. Active damping of a piezoelectric crystal designed for airborne acoustic ranging. Christopher L. Kenney and Ilene J. Busch-Vishniac (Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712)

Piezoelectric crystal ultrasonic transducers are ideal for airborne acoustic ranging because they are low-damping (high Q) devices capable of producing relatively high sound intensities at a single frequency. However, the low internal damping of the crystal results in substantial ringing, which can interfere with the reflected acoustic signal if left untreated. The conventional means of dealing with the ringing is to passively damp the crystal, but this causes a reduction in efficiency, increases the rise time, and adds substantially to the mass and cost of the transducer. An alternative approach, demonstrated here, is to actively damp the transducer using feedback control. In this system, the piezoelectric disk element is subdivided into two pieces: one large and one small. The small portion is used to sense motions resulting from driving the larger segment. The signal from the sensor is fed through a triggered classical proportional-derivative controller. The controller is triggered so that it is only active when it is desired to quench the crystal motions. Results show that using this controller it is possible to dramatically increase the damping of both the radial mode and the longitudinal mode of the crystal without adding mass to the system, decreasing the transducer operating efficiency, or increasing the rise time. [Work supported by General Motors Research Laboratories.]

11:00

LL13. Large area sensors for active acoustic control systems. Thomas R. Howarth, Xiaoqi Q. Bao, Vijay K. Varadan, and Vasundara V. Varadan (Department of Engineering Science and Mechanics, Research Center for the Engineering of Electronic and Acoustic Materials, The Pennsylvania State University, University Park, PA 16802)

The design and development of a composite piezoelectric polymer (PVDF) sensor for use in an active acoustic control system is presented. Large area PVDF plates are encapsulated in a polymer matrix to detect both the incident and reflected signals in an underwater pulse tube at low frequencies. These signals are amplified, phase shifted, and then used to drive an absorbing transducer to negate reflections. The results of this system are compared with earlier data [X.-Q. Bao *et al.*, *J. Acoust. Soc. Am. Suppl.* **184**, S49 (1988)] that utilized piezoceramic spherical hydrophones.

11:15

LL14. A theoretical basis for acoustic sensitivity and optimization analyses of three-dimensional radiators. Kenneth A. Cunefare and Gary Koopmann (Noise Control Laboratory, 157 Hammond, The Pennsylvania State University, University Park, PA 16802)

Numerical techniques have been developed that enable acoustic sensitivity and optimization analyses to be performed on three-dimensional

radiators. Sensitivity analysis of a body vibrating at a single frequency with a known normal velocity distribution indicates those areas of the surface that have the greatest influence upon the radiated power at that frequency. Regions of high sensitivity would be likely candidates for the application of damping materials or as locations for active control sources. Optimization analysis provides the optimum magnitude and phase at which to drive active cancellation sources in order to minimize the radiated power. Both techniques are based upon a discretized expression for radiated power, where surface pressure has been replaced by an expression that relates pressure to surface velocity through a transfer matrix. The transfer matrix is derived from a numerical discretization of the Helmholtz integral equation.

11:30

LL15. A new type of Gaussian ultrasonic transducer. Gonghuan Du, Desheng Ding, and Changming Gan (Institute of Acoustics, Nanjing University, Nanjing, People's Republic of China)

A Gaussian ultrasonic field is known to have some advantages [G. Du and M. A. Breazeale, *J. Acoust. Soc. Am.* **78**, 2083–2086 (1985)]. In this article, a new type of Gaussian ultrasonic transducer is presented with a curved back electrode instead of a usual plane electrode. The mathematics of design is based on an inverse approach. The theoretical result shows that the shape of the curved back electrode is dependent on the thickness of the transducer and on the Gaussian coefficient of vibration displacement on the surface of the transducer as well as on the ratio of dielectric constants of the transducer to the material of the back electrode. The inspection of displacement distribution is carried out on the surface of the transducer by using a laser interference technique. The measured results indicate good agreement with the prediction of the theory.

11:45

LL16. Characteristics of variable-reluctance transducers as underwater acoustic projectors. Elizabeth A. McLaughlin and Mark B. Moffett (Code 3111, Naval Underwater Systems Center, New London, CT 06320)

Variable-reluctance projectors, because they can deliver higher forces than moving-coil projectors, have long been proposed for underwater use. Single elements are nonlinear, however, and push-pull configurations of element pairs [W. T. Harris, U. S. Pat. No. 2,713,127; J. Chervenak, U. S. Pat. No. 3,725,856; G. Pida, U. S. Pat. No. 3,691,515; F. Massa, Jr., U. S. Pat. No. 3,319,220] have been used to eliminate second-order nonlinearities. Such push-pull arrangements are usually dipole radiators in nature and need to be baffled for monopole radiation. The displacement is normally limited by the gap thickness, but transverse variable reluctance devices [F. R. Abbott, U. S. Pat. Nos. 3,126,520, 3,353,040, and 3,605,080; J. Chervenak, U. S. Pat. No. 3,517,279] are not subject to this limitation and hence can produce greater displacements, albeit smaller forces, than can direct variable-reluctance transducers. The characteristics of these transverse variable-reluctance devices turn out to be similar to moving-coil projectors in many respects.

Session MM. Physical Acoustics VII: Reflection, Transmission, Dispersion, and Scattering

Anthony A. Atchley, Chairman

*Physics Department, Naval Postgraduate School, Monterey, California 93943***Contributed Papers****8:00**

MM1. Reflection of focused sound from curved, rigid surfaces. Michalakakis A. Averkiou and Mark F. Hamilton (Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1063)

Target geometry and orientation significantly affect the current accuracy of ultrasonic sensors used for gauging and proximity sensing [R. Hickling and S. P. Marin, *J. Acoust. Soc. Am.* **79**, 1151 (1986)]. The reflection of focused sound from curved, rigid surfaces using Gaussian beam solutions obtained from the parabolic approximation of the wave equation has been investigated. Both the focusing of the source and the curvature of the target are modeled by quadratic phase factors. A simple closed-form solution is thus derived for the sound reflected from concave and convex targets in either axisymmetric or nonaxisymmetric configurations. The validity of the solution is restricted to cases in which the focusing gain of the source, and the curvature of nonperpendicularity of the target, are not too large. Propagation curves, beam patterns, and scattering trajectories are used to illustrate the combined effects of diffraction and target curvature on the reflected sound field. Transfer functions are obtained by comparing the received to the transmitted acoustic power for the case of a monostatic acoustic sensor. [Work supported by GM and NSF.]

8:15

MM2. Echo identification using phase information. Nurgun Erdol (Department of Electrical Engineering, Florida Atlantic University, Boca Raton, FL 33431), Louis Roemer, Nathan Ida, and Ke-Sheng Huo (Department of Electrical Engineering, The University of Akron, Akron, OH 44325)

The signal waveforms chosen for reflectometry often result in overlapping echo waveforms. The waveforms chosen are constrained by transducer limitations, attenuation, and resolution properties of the test medium. The extraction of echo location information can be achieved using phase information, via the maximum entropy method. The phase estimate is constructed from the power spectrum of the even components (S_{ee}) of the signal and the power spectrum of the odd components (S_{oo}) of the signal. The cosine of the phase is

$$\cos[2\theta] = \frac{S_{ee}(e^{j\omega}) - S_{oo}(e^{j\omega})}{S_{ee}(e^{j\omega}) + S_{oo}(e^{j\omega})}.$$

By computing the spectrum of the above relation, the echo delay t_d can be computed using $\cos[2\theta(\omega)] = \cos[2\omega(t_d)]$ in the frequency domain. The method of analysis is presented, along with experimental verification that overlapped echoes are easily identified for a simple reflecting structure using ultrasound.

8:30

MM3. Transmission and reflection of a real sound beam at a two-fluid interface. Jacqueline Naze Tjøtta (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029, and Department of Mathematics, The University of Bergen, 5007 Bergen, Norway), Hans-

Christen Salvesen (Department of Mathematics, The University of Bergen, 5007 Bergen, Norway), and Sigve Tjøtta (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029, and Department of Mathematics, The University of Bergen, 5007 Bergen, Norway)

The analytical solution of many problems of acoustics is expressed in terms of a Fourier integral. In this paper, general asymptotic expressions that approximate both the location and the value of the maximum amplitude of a one- or two-dimensional Fourier integral are presented. These expressions are generally easier to compute than evaluations using a fast Fourier transform algorithm. The method is used to study the reflection and transmission of a real sound beam at the interface between two homogeneous and dissipative fluid layers. Simplified analytical formulas that explain the beam displacement and other behavior near the interface are obtained. Numerical results are compared with those of an earlier work [see Sagen, Naze Tjøtta, and Tjøtta, *J. Acoust. Soc. Am.* **85**, 24-38 (1989)]. The influence of various parameters on the direction and shape of the reflected and transmitted sound beams is investigated. [Work supported by the IR&D program of ARL:UT, The Royal Norwegian Council for Scientific and Industrial Research (NTNF), and VISTA/STATOIL, Norway.]

8:45

MM4. Reflection and transmission of elastic waves from a fluid/porous solid interface. Kunyu Wu, Qiang Xue, and Laszlo Adler (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210)

In this paper, Biot's theory of wave propagation in a porous solid has been applied to a theoretical study of energy reflection and transmission of elastic waves at oblique incidence on an interface between fluid and fluid-saturated porous solid. For this purpose, the necessary formalism of energy equation, Poynting energy flux vector, and sound intensity of elastic wave in fluid-saturated porous solid are presented. Two general cases of mode conversion have been investigated: (i) The initial wave is incident from the fluid to the interface and generates three transmitted bulk waves in the fluid-saturated porous solid, and (ii) the initial wave is incident from the fluid-saturated porous solid to the interface and generates three reflected bulk waves in the same media. The calculated results are in agreement with the law of conservation of energy. Furthermore, the transmission of sound through a fluid-saturated porous solid plate immersed in fluid is calculated to specify the optional condition supporting a strong slow compressional wave and to interpret Plona's experimental results. [This work was supported by the U. S. Department of Energy, Basic Energy Science Grant ED-FG02-84ER45057.A0005.]

9:00

MM5. Dispersive guided waves in interface layers. Peter B. Nagy and Laszlo Adler (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210)

It was recently suggested by the authors [*J. Acoust. Soc. Am. Suppl.* **1** **83**, S78 (1988)] that guided waves in an interface layer can be used to

characterize bond quality in adhesive joints. Experimental results clearly indicate that leaky guided modes of the interface layer itself are much more sensitive to most types of bond defects than the corresponding Lamb modes of the multilayered joint as a whole. Quantitative evaluation of the experimental results requires the development of an analytical technique to calculate the dispersion curves of guided waves in a solid (adhesive) layer between two solid (adherent) half-spaces. Special emphasis was placed on complex modes that are substantially attenuated by leaking into the substrates because of the relative easiness of generating and detecting them by simple experimental techniques. Differences between free and forced vibrations of the interface layer are considered, too. Experimental results are shown to be in good agreement with the calculated theoretical dispersion curves.

9:15

MM6. Study of the dispersion of the Scholte-Stoneley waves propagating at a plane interface. M. de Billy, G. Quentin (Groupe de Physique des Solides de l'ENS, University Paris 7, Paris, France), H. Franklin, M. Rousseau, and Ph. Gatignol (University Paris 6, Paris, France)

Theoretical results [Osborne and Hart, *J. Acoust. Soc. Am.* **17**, 100 (1945)] pointed out that for low values of the product Fd (frequency times the thickness of the plate), there exists a dispersion of the Scholte-Stoneley wave as it propagates along the interface between a liquid and a plate. Measurements of the velocity on different materials and for various thickness of the plate reveal a very small dispersion for values of Fd that ranges from 0.5–3 MHz mm. The experimental results will be compared with the theoretical calculations. [Work supported by DRET.]

9:30

MM7. Acoustic scattering from an air-filled prolate cylinder. G. Maze, F. Lecroq, J. L. Izbicki, J. Ripoche (Laboratoire d'Electronique et d'Automatique, Ultrasons, Université du Havre, Place Robert Schuman, 76610 Le Havre, France), and S. Numrich (Code 5133, U.S. Naval Research Laboratory, Washington DC 20375-5000)

The ultrasonic scattering from circular cylinders is described in numerous papers. If the insonification is perpendicular to the cylinder axis, the scattering is strongly influenced by the propagation of circumferential waves. These circumferential waves propagate either in the elastic shell, where they are the Whispering Gallery waves, or at the interface water/elastic shell, where they are the Scholte-Stoneley waves. These waves, for some frequencies called resonances, form standing waves around the cylinder. Previously, it has been shown that the resonance frequencies are functions of the cylinder diameter, the shell thickness, and the target composition. In this presentation, the influence of the curvature radius on the position and the amplitude of resonances is studied. The method of isolation and identification of resonances (MIIR) is used to plot resonance spectra of air-filled prolate cylinders. The incident ultrasonic beam perpendicular to the cylinder axis insonifies the cross section at different positions. The resonance spectra of an air-filled prolate cylinder are compared to those obtained with a circular cylinder that has the same thickness and the same circumference.

9:45

MM8. Acoustic signatures of passive and active targets by the MIIR. Jean Ripoche, Gérard Maze, Jean-Louis Izbicki, and Pascal Pareige (Laboratoire d'Electronique et d'Automatique, Ultrasons, Université du Havre, Place Robert Schuman, 76610, Le Havre, France)

The acoustic signatures of elastic targets immersed in water or inclusions in materials are more easily obtained and interpreted since the discovery and implementation, in the 1980s, of the "Method of Isolation and

Identification of Resonances" (MIIR) [G. Maze and J. Ripoche, *Phys. Lett. A* **84**, 309–312 (1981); *J. Acoust. Soc. Am.* **73**, 41–43 (1983)] and derived pulse methods. The first interest is the isolation of resonances in quasiline spectra, which are obtained for passive or active targets when the excitation is external or internal. The second interest is the experimental identification of each resonance by the determination of the n mode number, which allows a classification of resonances. The results corroborate the "Resonance Scattering Theory" [H. Überall, L. R. Dragonette, and L. Flax, *J. Acoust. Soc. Am.* **61**, 711–715 (1977)]. Quasi-harmonic MIIR and short-pulse MIIR can be applied to cylinders, air- or liquid-filled shells, and multilayered structures (or other targets) immersed in water by means of an external or internal excitation. Experiments give resonance spectra of liquid cylindrical inclusions in elastic materials. [Work supported by D.R.E.T., D.G.A., Contract 86/045.]

10:00

MM9. Scattering from elastic shells in time domain at coincident frequencies. M. F. Werby and H. Ali (NORDA Numerical Modeling, SCC, MS 39529)

Scattering from the submerged elastic spherical shells at the frequency at which the phase velocity of the flexural (A_0) mode is approximately equal to the speed of sound in the ambient fluid is considered. This is referred to as the coincident frequency for plates, and it produces strong flexural vibrations in both plates and spherical shells. The resonances that arise for bounded objects are observed to have a fairly constant group velocity and resonance width over a broad ka region. It is shown that this is sufficient to produce a large backscattered return of a characteristic shape in the time domain. This is demonstrated for several examples.

10:15

MM10. The Rayleigh and flexural resonances for elastic solids and shells. M. F. Werby (NORDA, Numerical Modeling, SCC, MS 39529) and G. C. Gaunard (NSWC, White Oak, Silver Spring, MD 20910)

In a comparative study, the Rayleigh (leaky Rayleigh-type) resonance and the flexural resonances for the solid and shells of thicknesses of 1% to 95% of the outer radii are calculated. It is shown that, as the shell thickness increases, the flexural resonance goes into the Rayleigh resonance at about a thickness of 60%. Further, these resonances are observed out to a ka of 500. It is also shown that thin shell theories predict the resonance shapes or the flexural modes for the thin cases, including the well-known effect of an upturn at the lower end of the frequency range.

10:30

MM11. A suitable background for elastic shells. M. F. Werby (NORDA Numerical Modeling, SCC, MS 39529)

It has been known for some time that a useful background in the description of scattering from elastic solids is the rigid scatterer. When resonances are not too strongly overlapping, then, by subtracting a rigid background, it is possible to isolate the resonance response from the elastic solid. Previously, it was reported that, for very thin shells at low frequency, a soft background was suitable and that, at very high frequencies for thicker shells, a rigid background was suitable. A suitable background for the elastic shell for all frequencies and for shells to moderate thickness is derived, and its utility is demonstrated with several examples.

MM12. The acoustic scattering by a submerged, elastic spherical shell: High-frequency limit. Roger H. Hackman and Gary S. Sammelmann (Naval Coastal Systems Center, Physical Acoustics Branch (Code 2120), Naval Coastal Systems Center, Panama City, FL 32407-5000)

This presentation represents the third in a series devoted to the analysis of the analytic structure of the acoustic scattering amplitude of an elastic spherical shell. In previous meetings [J. Acoust. Soc. Am. Suppl. 1 **83**, S94 (1988); **84**, S185 (1988)], a fundamentally oriented analysis of the pole structure of the scattering amplitude in the low- to mid-frequency region was presented. The most surprising result of this investigation was the demonstration that the fluid-loaded antisymmetric Lamb wave bifurcated near the frequency that the vacuum dispersion curve transitioned from a subsonic to a supersonic phase velocity. In this presentation, the previous analysis of both the behavior of the acoustic scattering amplitude of a thin spherical shell and the pole structure of the S matrix is extended to the high-frequency region ($100 < ka < 1000$). The appearance of behavior that has a strong analogy with that of the higher-order flat plate Lamb waves is demonstrated. In particular, the appearance of strong thickness resonances, regions of anomalous dispersion, and regions of negative group velocity analogous to those observed by Tolstoy and Usdin [J. Acoust. Soc. Am. **29**, 37-42 (1957)] in their study of the dispersion properties of higher modes on a plate is discussed.

11:00

MM13. An analysis of the modes of an elastic prolate spheroid: From a spherical to cylindrical geometry. Gary S. Sammelmann and Roger H. Hackman (Naval Coastal Systems Center, Physical Acoustics Branch (Code 2120), Panama City, FL 32407-5000)

The acoustic scattering from a solid prolate spheroid is studied as a function of frequency and aspect ratio. The emphasis of this study is on the nature and the coupling of the elastic excitations of a solid prolate spheroid in the transition from a spherical to what is essentially a cylindrical geometry. The aspect ratio of the spheroid from the spherical limit to the almost cylindrical limit is varied. In this manner, it is possible to classify the elastic excitations of low-aspect ratio prolate spheroids in terms of the elastic modes of vibration of a sphere and to study how these modes undergo transformation to the modes of a cylinder with respect to increasing aspect ratio. A qualitative comparison of the behavior of these low-frequency elastic excitations of a prolate spheroid and their quasicylindrical mode interpretation is made.

11:15

MM14. Total scattering cross section of an elastic spherical shell: Comparison of exact computations with a GTD model that includes Lamb wave resonances. Steven G. Kargl and Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164-2814)

The optical theorem, relating the total scattering cross section $\sigma_t(ka)$ to the forward scattering function $f(\theta=0, ka)$, is used to study the rich structure contained in $\sigma_t(ka)$ for an elastic spherical shell in the frequen-

cy range $7 < ka < 100$. The partial-wave series (PWS) for $f(0, ka)$ does not facilitate a direct understanding of the physical mechanisms causing this structure. Hence, a model for $f(0, ka)$ is developed from an elastic generalization of the geometrical theory of diffraction (GTD) [P. L. Marston, J. Acoust. Soc. Am. **83**, 25-37 (1988)]. The generalized GTD is based on the Watson transformation of the PWS. The GTD model includes explicit contributions from ordinary forward diffraction and from individual Lamb waves excited on the shell. Calculations of $\sigma_t(ka)$ with $f^{\text{PWS}}(0, ka)$ and $f^{\text{GTD}}(0, ka)$ show good agreement between the exact PWS result and the GTD model for the ka range investigated. The agreement confirms the GTD model and demonstrates the relevance of Lamb waves guided by the shell and forward glory scattering. Finally, these calculations verify the correctness of the numerical methods by which the Lamb wave parameters were obtained [S. G. Kargl and P. L. Marston, J. Acoust. Soc. Am. **85**, 1014-1028 (1989)]. The numerical computations presented are for a stainless steel shell with an inner-to-outer radius ratio $b/a = 0.838$. [Work supported by ONR.]

11:30

MM15. An application of the forward-scattering theorem to elastic wave attenuation in inhomogeneous materials. Kurt P. Scharnhorst (Naval Surface Warfare Center, White Oak, Silver Spring, MD 20903-5000), Roger H. Hackman, and Raymond Lim (Naval Coastal Systems Center, Panama City, FL 32407)

The forward-scattering amplitude of two adjacent scatterers $f(2)$ may be used to construct an effective medium theory of inhomogeneous media containing random distributions of scatterers. The amplitude $f(2)$ may be thought of as the lowest-order approximation of the amplitude $f(n)$, of an interacting cluster of n scatterers contributing to the formation of the coherent elastic wave. Since the elastic wave encounters all orientations of pairs, the averaged nearest-neighbor forward-scattering amplitude is studied. Specific orientations that are dictated by the dynamics of the scattering processes are also considered. The average forward-scattering amplitude is analyzed in terms of the direct σ_{nn} mode conversion, σ_{nn} , and absorption cross sections for random distributions of viscoelastic spherical scatterers in elastic matrix materials. Estimates of longitudinal wave attenuation due to these three mechanisms in specific materials are presented.

11:45

MM16. Acoustic sampling volume. K. G. Foote (Institute of Marine Research, 5024 Bergen, Norway)

Knowledge of the sampling volume is necessary in many quantitative applications of acoustics. In general, the sampling volume is not merely a characteristic of the transmitting and receiving transducer or transducers, but also depends on the concentration and scattering properties of the target, the kind of signal processing performed on the echo, and the detection threshold. These dependences are stated explicitly in formulas for the sampling volume and a differential measure, the effective equivalent beam angle. Numerical examples are given for dispersed and dense concentrations of both point scatterers and directional fish scatterers.

Session NN. Speech Communication VII: Fundamental Frequency, Timing, and Stress in Words and Sentences

Robert F. Port, Chairman

Department of Linguistics, Indiana University, Bloomington, Indiana 47405

Contributed Papers

8:00

NN1. Articulatory correlates of stress clash rhythms. Mary E. Beckman (Ohio State University, Department of Linguistics, 204 Cunz Hall, Columbus, OH 43210-1229)

Linguistic characterizations of English stress patterns typically claim that when one stressed syllable is followed immediately by another and the rhythmic clash is not corrected by retracting the first stress to an earlier position, then the first vowel will be protracted, increasing the phonetic separation between the adjacent beats and thereby regularizing the rhythm. Acoustic studies, however, consistently show that the mean duration for the first vowel in such a stress clash is not substantially longer than that of its counterpart in an alternating stress pattern. Examination of articulatory correlates suggests an explanation for this discrepancy: Syllables in a stress clash may undergo a rhythmic reorganization of segmental gestures that can distance the first syllable's prosodic peak from the following stress without lengthening the vowel's measured duration. Displacements, durations, and peak velocities of jaw opening and closing gestures were measured for four speakers' productions of stress clash sequences *Pop posed* and alternating stress sequences *Pop opposed*. In general, the jaw opening gesture was relatively shorter and the closing gesture relatively longer in the stress clash context. If the degree of jaw opening is taken as an approximation to the sonority contour within the syllable, this result means that the prosodic peak occurred relatively earlier within the initial [pap] in *Pop posed*. [Work supported by the National Science Foundation under grants IRI-8617852 to Mary Beckman and IRI-8617873 to Jan Edwards.]

8:12

NN2. Stress clash in isolated phrases and sentence contexts. Michael S. Cluff and Keith Johnson (Department of Psychology, Indiana University, Bloomington, IN 47405)

The experiment reported in this paper investigates the possibility that stress clash should be called "accent clash." Stress clash in the phonological literature has been defined in terms of lexically specified stresses, while experimental investigation of such stress clash candidates has given a mixed picture as regards any phonetic effect of adjacent lexical stresses. The hypothesis was that lexical stress defines the candidates for a clash and that the placement of accents during production determines whether a clash will actually occur. To test this, subjects' productions of pairs of adjective noun phrases that contrasted stress clash versus no stress clash environments (big dinosaur/big dimension) were analyzed. These phrases were recorded both in isolation and within syntactically identical sentence environments. This manipulation was used to elicit two types of accent patterns; in isolation, both the adjective and noun received intonational accent, while in context, only the noun was accented. The results of a pilot experiment have suggested that, for phrases of this type, there is a difference in the durational effect of stress clash (duration of the adjective) when items are produced in isolation versus when they are produced in sentence contexts. The relevance of these data for current phonological theory will be discussed. [Work supported by NIH Research Grant NS-12179 to Indiana University.]

8:24

NN3. Effects of the mora and segmental features on word duration in Japanese. Robert Port and Jane Hardy (Department of Linguistics, Indiana University, Bloomington, IN 47405)

Word duration in Japanese is determined by number of moras when words are pronounced at a normal speaking tempo in focus position in carrier sentences [Port *et al.*, J. Acoust. Soc. Am. **81**, 1547-1585 (1987)]. Still, variation in segmental makeup has some effect on word duration. It was expected that, when out of focus position or when spoken at faster tempo, durations for words with a given number of moras would show more variation. A set of Japanese words of two, three, and four moras having various syllabic templates (e.g., *tokki*, *tookki*, etc.) was chosen. They were produced in both focus position and nonfocus position (by changing a nontarget word from sentence to sentence) at two tempos. Words with same mora length exhibited less word-to-word variation at slow tempo than fast, and less variation in the focus position than when nonfocused. To factor out segmental effects, subjects were asked to produce each sentence reiterantly by replacing each mora of the target word with *ta*. As expected, reiterant results showed almost no variation due to original segments and much less effect of sentence position and tempo. The results suggest that competing constraints affect Japanese timing: a mora constraint ("make all moras equal") plus intrinsic segmental constraints (different for each segment type). Mora timing dominates at slower tempos and in focus position, but can be overpowered by constraints of intrinsic segmental duration elsewhere. [Work supported by NSF, DCR-85-18725.]

8:36

NN4. A cross linguistic contrast in the temporal compensation effect. Takashi Otake (Department of General Education, International Budo University, 841 Shinkan, Katsuura, Chiba 299-52 Japan)

The purpose of this investigation is to evaluate whether the temporal compensation effect in Japanese can be attributable to mora timing. It has been asserted that the temporal compensation effect has an important role in regulating a CV syllable duration in Japanese [Port *et al.*, *Phonetica* **37**, 235-252 (1980)]. In an earlier study [T. Otake, J. Acoust. Soc. Am. Suppl. **1** **84**, S97 (1988)], Arabic and Japanese were investigated under the same conditions with respect to the compensation effect, and it was found that both languages showed the same compensation effect, which may suggest that it is a universal phenomenon [Beckman, *Phonetica* **39**, 113-135 (1982)]. The experiment reported here used the method of Port *et al.* (1980) to test the hypothesis further by investigating other languages that belong to stress timing (English and German) and syllable timing (Spanish and French). In addition, Chinese, which does not belong to any of these timings, was investigated. Pilot results show that the temporal compensation effect reported by Port *et al.* (1980) can be equally observable in the above languages.

8:48

NN5. On the temporal alignment of F0 with segmental structure. Kim E. A. Silverman (Room 2C-440, AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974)

Timing and F_0 tend to be correlated in speech. For example, a syllable near the end of a phrase will be lengthened, and if it bears a local F_0 maximum (a high pitch accent), then the peak will occur earlier in that syllable. The more the lengthening, the earlier the accent peak. However, the current experiment shows a case where F_0 peaks are aligned earlier in their syllables without the expected concomitant lengthening. Two speakers recorded *My ___ lives in Memphis*, where the second word was *Ma*, *Mom*, *Mama*, *Pa*, *Pop*, or *Papa*, both with and without a low F_0 target (corresponding to an utterance-internal prosodic boundary) after the second word. Acoustic measurements, combined with perceptual judgments, show that the low tone pushes the preceding pitch accent about 50% earlier in its syllable, with little or no lengthening of that syllable. Thus peak alignment depends on upcoming tonal events, as well as those utterance features that determine durational structure.

9:00

NN6. Prosodic aspects of French speech rhythm. Janet Fletcher (Department of Linguistics, Ohio State University, 204 Cunz Hall, Columbus, OH 43210-1229)

9:12-9:22

Break

9:22

NN7. Evidence for levels of prosodic hierarchies in Swedish. J. M. Rauschenberg (Department of Linguistics, Ohio State University, 204 Cunz Hall, 1841 Millikin Road, Columbus, OH 43210)

Pierrehumbert and her colleagues have proposed a model of prosodic organization that utilizes not only types of constituents and their marker(s), but also well-defined edges. Beckman and Pierrehumbert (1986) have suggested that, in English, the phrase tone following the nuclear accent marks the right edge of an intermediate phrase (ip), giving evidence of phrasing internal to the intonational phrase (IP). Beckman and Edwards (1987) account for a final lengthening effect in English as one marker of the edge of the IP. Swedish differs from English in that sentence accent does not occur at a phrase edge. Thus, if there is a level at which there is one and only one sentence accent, utterances with single terminal junctures and multiple phrase tones must be analyzed as having one IP with multiple ip's. But there is no tonal marker for the edges of the ip's. An experiment has been constructed (in progress) to test the hypothesis that lengthening will occur at the end of the IP and, to a lesser extent, the ip, and that this might allow a determination of edges of Swedish phrasal units. Thus words at the ends of phrases should be longer than in medial position. The corpus consists of sentences, each having five accent 2 words, in six different constructions. The predictions are, specifically, (1) that the test word will be longer in sentence final position, and (2) that in tokens that show internal phrasing (as indicated by multiple phrase/sentence tones), the word will be longer where it occurs at the edge of an internal constituent than where medial to a single phrase constituent.

9:34

NN8. Relation between the features "givenness" and "accentedness" and the duration of Dutch words. Wieke Eefting (Institute of Phonetics, University of Utrecht, Trans 10, 3512 JK Utrecht, The Netherlands)

A major assumption of the syllable-timing/stress-timing distinction is that interstress interval length in a language like French is a direct additive function of the number of syllables it contains. In stress-timed languages like English, by contrast, interstress interval length is supposed to show a relationship of negative acceleration with increasing syllable number due to an underlying tendency towards the isochronous spacing of stresses. Nakatani *et al.* (1981), however, showed that interstress intervals are not only positively correlated with intervening number of syllables, there is also no relationship of negative acceleration between foot size in syllables and foot duration. Acoustic timing data for French concur with Nakatani *et al.*'s findings for English. Accentual phrase duration was highly correlated with the number of syllables that unit contained. Although disyllabic units were not twice as long as monosyllabic units in these French data, this could reflect the fact that rhythmic units in French include a long final syllable due to a right boundary accent. The similarity of these results and those for American English would suggest that apparent compression effects are not a strong argument in favor of one typological description as opposed to another. A further experiment revealed lengthening patterns associated with phrasing that are hierarchical in nature and are compatible with recent models of intonational phrasing proposed for French (e.g., Martin, 1987) and highlight the relevance of accentual grouping in descriptions of French timing.

The aim of this study is to investigate whether the duration of a word in speech production is related with the information value (carrying "new" or "old" information) and with the accentedness (+ or - accent) of the word. Fowler and Housum [C. A. Fowler and J. Housum, *J. Mem. Lang.* 26, 489-504 (1987)] found that speakers tend to shorten words conveying old information. It has also been found, by others, that accented words have a longer duration than words that do not carry accent. Because the features new/accented and old/unaccented appear to be closely related, it is unclear which of the two features causes the differences in duration as found by Fowler and Housum. In the present study, it is attempted to sort out the effects of accentedness and information value. Data will be presented showing how both factors separately and combined affect word durations.

9:46

NN9. Lexical neighborhoods in speech production: A first report. Stephen D. Goldinger and W. Van Summers (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Investigations of speech production have shown that talkers will systematically alter the acoustic-phonetic properties of their utterances in response to changes in the context in which the words are spoken. Well-known examples of such contexts are the presence of a loud background noise [e.g., Lombard (1911)], or the linguistic context surrounding the target word in a sentence [e.g., Lieberman, *Lang. Speech* 6, 172-188 (1963)]. Recent work by Balota and Shields [Psychonomic Soc. Conf. (1988)] suggests that factors intrinsic to words, such as their frequencies, may also affect the durations of spoken words. The present paper reports the results of a preliminary investigation of the effects of similarity neighborhood structure on speech production. Global, as well as segmental, comparisons of subjects' productions of words from dense and sparse lexical neighborhoods will be presented. [Research supported by NIH research grant NS-12179-11.]

NN10. An articulatory characterization of contrastive emphasis in correcting answers. John R. Westbury (Waisman Center, University of Wisconsin, Madison, WI 53705-2280) and Osamu Fujimura (Speech and Hearing Science, Ohio State University, Columbus, OH 43210-1372)

New recordings of articulatory movements of the tongue, lips, and mandible have been obtained using the University of Wisconsin x-ray microbeam system, from eight normal adult speakers of American English who uttered three-number sequences in question-answer pairs such as "Is it 995 Pine Street? No, it's 955 Pine Street." Analysis of these data is intended to provide a detailed characterization of the implementation of contrastive emphasis observed in this context. Preliminary comparisons of maximum displacements and peak velocities for the tongue blade and lower lip, during the words "nine" and "five," respectively, show that both movement parameters increase during a syllable under the influence of contrastive emphasis, but that the relative change in articulator displacement is always greater. This observation is at least partly consistent with results of previous research [Fujimura and Spencer, *J. Acoust. Soc. Am. Suppl.* 1 74, S117 (1983)], which showed that maximum articulator displacements during emphasized and unemphasized vowels differed more than did certain portions of the movements toward and away from those vowels. [Research supported by USPHS Grant NS-16373.]

10:10

NN11. Acoustic characteristics of contrastive stress production in normal geriatric, apraxic, conduction aphasic, and dysarthric speakers. Julie M. Liss and Gary Weismer (Department of Communicative Disorders and Waisman Center, University of Wisconsin-Madison, Madison, WI 53705)

Contrastive stress drills are often used in speech therapy for persons with neurogenic speech disorders. These drills are designed to have the patient produce heavy stress on a word embedded in a sentence, presum-

ably to maximize articulatory displacement (and perhaps derivatives or results of displacement, such as articulatory velocity or contract). Like many manipulations used in speech therapy, there are no data indicating the efficacy of the technique and, at a more basic level, little data that bear on the phenomena resulting from application of the technique. The purpose of this paper is to describe how vocalic formant trajectories and segmental durations vary as a function of contrastive stress in four groups of speakers. The groups include normal geriatric persons, persons with apraxia of speech, ataxic dysarthria, and conduction aphasia. Results will be discussed relative to the underlying articulatory behavior that is affected by the stress manipulation.

10:22

NN12. Effects of cognitive workload on speech production. W. Van Summers, David B. Pisoni, and Michael A. Stokes (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Previous research on the acoustic properties of speech has generally focused on "normal" speech produced under benign conditions. Typically, subjects produce utterances in a quiet laboratory setting with no competing or distracting stimuli present. Almost no basic research has examined speech in more demanding environments. For example, pilots may often be required to produce speech while simultaneously performing several other attention-demanding tasks. The information-processing requirements of these simultaneous tasks may affect the acoustic properties of speech in a wide variety of ways. The present research examined speech produced under benign conditions and while simultaneously performing an attention-demanding perceptual-motor task. Speech produced while carrying out the perceptual-motor task showed reliable increases in amplitude and fundamental frequency and reductions in segmental durations and spectral tilt. The results have implications for automatic speech recognition in demanding environments. [Work supported by Armstrong Aerospace Research Laboratory, Wright-Patterson AFB, Contract No. AF-F-33625-86-C-0549.]

10:34-10:44
Break

10:44

NN13. Initial prominence and accent in Seoul Korean prosody. Kenneth de Jong (Department of Linguistics, Ohio State University, Columbus, OH 43210)

This paper investigates two questions about the prosodic system of Seoul Korean—whether there is a word-initial prominence, and whether a concurrent F_0 peak is best seen as an optional pitch accent or as a boundary tone associated with some prosodic level. It analyzes the utterances of two speakers of the Seoul dialect. The durations of syllables and vowels placed in various positions within a sentence are compared. The two intonational analyses are tested by noting the position of the F_0 peak, determining how consistently the peak appears, and noting the effects of pitch range variation. Results suggest: (1) that initial syllables of a word are consistently longer than medial syllables; (2) the position of the initial F_0 peak is closely bound to the initial syllable of the utterance, but not of the word; and (3) the initial F_0 peak appears in all tokens, except relatively short or fast tokens. This lack is shown to be due to tonal undershoot of a following low tone, as is apparent from strong and continuous relationships between height of pitch fall and rise and the temporal distance between the F_0 peaks. These results suggest there is a word initial prominence in Seoul prosody, distinct from initial tones. The tones are best analyzed as boundary markers for some larger prosodic level, which become associated with initial syllables.

10:56

NN14. The accentual pattern and prosody of the Chonnam dialect of Korean. Sun-Ah Jun (Department of Linguistics, Ohio State University, 204 Cunz Hall, 1841 Millikin Road, Columbus, OH 43210)

This paper examines the pitch accent and intonation patterns of the Chonnam dialect of Korean, spoken in the South Cholla province. The F_0 contours show that the Chonnam dialect has two kinds of accentual patterns, low-high-low and high-high-low, the choice of which is determined by the first segment of a phrase. If the segment has a laryngeal feature of either [+ spread] or [+ constricted], a phrase beginning with the segment gets a h-h-l pattern. Otherwise, it gets a l-h-l pattern. Cross linguistically, it is well known that aspirated or glottalized segments cause higher F_0 at the onset of the following vowel. However, these phonetic facts do not mean that the h-h-l pattern is a low-level product of the physiological effect. Rather, the occurrence of this accent pattern defines a phonological phrase. The F_0 contours also suggest that the Chonnam dialect has two more units of prosodic structure above the phonological phrase within an utterance, namely the intermediate phrase and the intonational phrase. An intermediate phrase in Chonnam is characterized as the domain of downstep between phonological phrases. An intonational phrase is characterized by a high or high-low boundary tone. The data used in this paper are from the dialect spoken in Kwangu, the main city in the Chonnam province.

NN15. Intonational domains of the Seoul dialect of Korean. Sookhyang Lee (Ohio State University, Department of Linguistics, 204 Cunz Hall, Columbus, OH 43210)

Pitch tracks were analyzed to determine the prosodic domains needed to describe intonation patterns in the Seoul dialect of Korean. Three different domains are observed: the intonation phrase, the accentual phrase, and the phonological word. The intonation phrase consists of one or more accentual phrases, and is cued by the occurrence of a final falling or rising boundary tone configuration. The accentual phrase consists of one or more phonological words and is cued by an F_0 peak at the beginning and a low-high accent on the last syllable. Since the initial peak is not associated with every word-initial syllable, it belongs to the accentual phrase and is not a correlate of word-initial lexical stress. Undershoot of the initial peak occurs when the accentual phrase is very short. There is also downstep between each accentual phrase within the intonational phrase; the pitch range is reset at the beginning of a new intonation phrase.

NN16. Prosodic structure in language understanding: Evidence from tone sandhi in Mandarin. Shari R. Speer (Department of Psychology, P. O. Box 11A Yale Station, New Haven, CT 06520) and Chi Lin Shi (Department of Modern Languages and Linguistics, Cornell University, Ithaca, NY 14853)

Two experiments show that listeners can use phrase-level prosodic information to disambiguate local lexical ambiguities that occur due to the operation of the tone sandhi rule in Standard Mandarin Chinese. In Chinese, each word is associated with a tone; changes in a word's tone that occur in relation to the tones of words nearby in phrasal context are referred to as "tone sandhi," e.g., ([L] → [LH]/_ [L]). In experiment 1, listeners identified lexical tones for ambiguous, unambiguous, and nonsense words in phrasal contexts where the tone sandhi rule might have applied. Comparable results in the lexical versus nonsense conditions indicate that judgments did not rely simply on lexically stored tonal information, but also made reference to prosodic structure. In experiment 2, subjects chose the correct written English translation for auditory sentences of Mandarin. Global prosodic information was manipulated to create different levels of "prosodic closeness" between two critical items in a tone sandhi environment, but the syntactic relation between these items was held constant. Results show that listeners relied on the prosodic structure of the phrases to determine whether or not the tone sandhi rule had applied, and consequently to identify individual lexical items. The evidence is taken to support the notion that prosodic structure influences auditory language comprehension processes. [Work supported by NICHHD.]

NN17. Phonetic implementation rules for tones in Yorùbá. Yetunde Laniran (DMLL, Morrill Hall, Cornell University, Ithaca, NY 14853)

This paper reports on some aspects of the phonetic implementation rules for tones in Yorùbá declarative sentences. Some of the current literature on intonation suggests that declination is not sensitive to the phonological composition of the tones of an utterance. In Yorùbá, tonal composition is relevant, since the rate at which pitch drops in identical tone sequences depends on the tone type. H and M sequences both begin at the same pitch level and are only distinguished subsequently by the degree of declination in the M sequence. The rate of declination in a sequence of L tones is fastest of all. The largest drop in pitch of L sequences is between the final and penultimate L tone for two of the three speakers we have considered so far. Another feature of the F_0 contour in Yorùbá is the phenomenon of high-raising by which a H tone is raised by a following and/or a preceding L tone. High-raising seems to be a cumulative effect of surrounding L tones, the pitch value of the H tones decreases in the following representation: LHL > LHM ~ MHL > MHM.

NN18. Normative and functional long-term acoustical aspects of English and French men's and women's continuous speech produced normally or with masking noise. William Weiss (University of Ottawa, Ottawa, Ontario K1N 6N5, Canada)

An acoustic measure of voice quality was proposed by Frokjaer-Jensen and Prytz [Tech. Rev. (1976)] as α = intensity above 1 kHz/intensity below 1 kHz. Sundberg and Gauffin, [Q.P.S.R. 2-3, (1978)] seemed to suggest that judging higher spectra as a measure of quality is misleading because it could be obtained with an increased vocal effort. They proposed that a measure of good quality is a higher increase of energy in the F_0 area relative to the F_1 area ("flow" phonation). The objective of the investigation was to find out how *tension*, *voice quality*, and *spectra levels* are affected under differential masked (that is, functional) conditions for (1) men and women, (2) anglophones and francophones. It was found that men have significantly higher F_1/F_0 than women and that Anglophones have significantly higher α than Francophones under masked conditions. However, in spite of the normative differences, the spectral changes were not significantly different between the groups. These results shall be discussed in view of existing theories of acoustic measurements of voice quality.

Session OO. Architectural Acoustics III and Musical Acoustics IV: Music Education Facilities Since 1975

Richard H. Talaske, Chairman

*The Talaske-Joiner Group, Inc., 137 North Oak Park Avenue, Oak Park, Illinois 60301***Chairman's Introduction—8:30***Invited Papers***8:35****OO1. The dynamics of a music education facility design team—The role of acoustical consultant.** Dennis A. Paoletti (Paoletti/Lewitz/Associates, Inc., 40 Gold Street, San Francisco, CA 94133)

The design of facilities for music performance, rehearsal, and education is a complex task. Some of the major aspects of realizing such facilities include: Teaming, marketing, negotiating contracts, establishing criteria, client/user interface, technical acoustical analysis and design, aesthetics, consulting, compromise, budgeting, construction/installation, contractor interface, acoustical measurements, acceptance, and evaluation of user and audience satisfaction. This paper will discuss the prominent role of the acoustical consultant in the process of designing facilities for music performance, rehearsal, and education.

9:00**OO2. Planning school and college music facilities.** Harold P. Geerdes (Geerdes Consulting Services, 2210 Woodlawn Avenue, S. E., Grand Rapids, MI 49506)

Educational music facility planning has become a specialty within the broad field of acoustics. While teaching, rehearsal, and performance spaces for school and college music departments share many of the requirements for downtown concert halls and opera houses, they are in many ways unique. A school or college music facility planner ideally combines a practicing background as a teacher and conductor with knowledge and experience in room acoustics, noise control, and electronic systems. As a member of the design team, he represents the music staff of the educational institution and articulates their needs to the other planners. Throughout the design development process he provides a level of expertise beyond that of the music educator. A close relationship with the 55 000 member Music Educators National Conference is very beneficial. The educational music facility planner can work under contract with the client or with the architect, depending on the particular circumstances, or he may work for a firm of established acoustical consultants who wish to use his special experience and expertise to supplement their own.

9:25**OO3. The acoustics of small rooms for music practice and teaching.** David L. Klepper (Klepper Marshall King Associates, Ltd., White Plains, NY 10603)

At best, the small rooms typical for music teaching and practice are unnatural environments for music production. Given the usual constraints of budget and practicality, a small room cannot be made to approximate the acoustics of a fine concert hall. If all normal modes are suppressed completely by sound-absorbing treatment, the room will be unnaturally dead or anechoic, and will be disliked by students and instructors. Yet, if normal modes are not controlled and the small room left "live," it will not sound like a normal concert hall but be "boxy," with a very uneven frequency response, and will be equally disliked. Compromise solutions involving fixed treatment on one wall or two adjacent walls, or adjustable treatment, with varying degrees of room irregularity and splaying, which have provided satisfaction in music teaching buildings, will be discussed. Organ practice rooms pose a separate problem, and solutions involving no applied sound-absorbing treatment, but considerable splaying or irregularities in ceiling and wall surfaces appear best. Some discussion of sound isolation will be presented.

OO4. The acoustic design of recital halls for educational facilities. J. Christopher Jaffe (Jaffe Acoustics, Inc., 114A Washington Street, Norwalk, CT 06854)

A recital hall for an educational facility requires a close appraisal of anticipated teaching, rehearsal, and performance use. In most cases, a modified set of acoustic criteria is required for these spaces in contrast to those associated with the design of professional recital halls such as Weill Hall, Town Hall, or the 92nd Street YMHA. This paper discusses the special requirements of recital halls in educational facilities, recognizes the work of Rein Pirn in this area, and develops a revised scale for volume-to-seating area ratios for these rooms. Qualitative subjective judgments related to the ratio of direct-to-reverberant energy are reviewed and measurements taken in some representative recital halls are analyzed.

10:15

OO5. Schnitger meets Sabine: Designing for organs. Robert F. Mahoney (Kirkegaard & Associates, 954 Pearl Street, Boulder, CO 80302)

In the planning of music education facilities there are few areas that afford the possibility of collaboration between the acoustician and the instrument builder; the design of spaces for organ practice and performance is one of these. The organ builder has within the traditions of his or her craft considerable latitude in regard to tonal design, pipe scaling, and the disposition of various divisions. The acoustician, on the other hand, must serve an organist who is likely to relish a highly diffuse and reverberant sound, but who is often confined to practice rooms of limited volume and is seated close to a powerful instrument of considerable physical extent. In addition to the customary concerns of adequate isolation and suitable background sound levels, the designer must marry the room acoustical design to the characteristics of the instrument to be installed there, recognizing the distinction between practice and performance environments.

10:40–10:45

Break

Contributed Papers

10:45

OO6. Acoustical considerations for two music instruction buildings. Roy L. Richards (Towns, Richards & Chaudiere, Inc., 105 N. E. 56th Street, Seattle, WA 98105)

The acoustical design features and performance data for music instruction and performance facilities at George Fox College in Newburg, Oregon (completed in 1982) and Whitman College in Walla Walla, Washington (completed in 1985) will be discussed. Since these facilities house high level activities that demand virtually no intrusion of sound from adjacent spaces, methods for achieving a high degree of sound isolation will be emphasized. At George Fox, the rehearsal rooms were located in a former gymnasium, thereby imposing design restrictions in addition to those dictated by a limited construction budget.

Light reflections can provide an easily understandable analogy to some acoustical features of architectural spaces. The interior surfaces of a 30-x 60-ft lecture/conference room were covered with reflective plastic. Reflections from various light sources were observed and photographed. Sound blasts were generated in the same room and the resulting reflections were recorded and analyzed. Results from the two experiments were compared in order to determine the nature and extent of the correspondence.

11:15

OO8. Optimizing home listening rooms. Peter D'Antonio (RPG Diffusor Systems, Inc., 12003 Wimpleton Street, Largo, MD 20772)

Various aspects of the listener, the loudspeaker, and the listening room will be discussed. The psychoacoustic mechanisms used by the auditory system are briefly reviewed to understand how the listening room can corrupt spatial and spectral textures. Experimental polar patterns, measured at the ear drum, for a human ear, a dynamic speaker, a dipole speaker, and an omnidirectional speaker, will be presented. TEF measurements will be used to demonstrate how the loudspeaker polar response dictates the type of acoustical treatment necessary. After a discussion of the principles behind the designs which comprise the RPG Home Concert Hall, a detailed outline of three listening room designs for dynamic speakers (Opus I), dipole speakers (Opus II), and spatially oriented or omnidirectional speakers (Opus III) will be presented. Experimental before and after TEF ETC measurements are presented and discussed.

11:00

OO7. Sound reflections and their optical counterparts. Julian L. Hook (School of Architecture, University of Illinois at Urbana-Champaign, 608 E. Lorado Taft Drive, Champaign, IL 61820), Arne S. Gullerud (University of Illinois, Urbana-Champaign, IL 61820), Paul D. Schomer (U. S. Army Construction Engineering Research Laboratory and University of Illinois Department of Electrical and Computer Engineering, Champaign, IL 61820), and Ingvar Schousboe (School of Architecture, University of Illinois at Urbana-Champaign, 608 E. Lorado Taft Drive, Champaign, IL 61820)

OO9. The significance of background sound in acoustical privacy for multifamily dwellings. Timothy J. Foulkes (Cavanaugh Tocci Associates, Inc., 327 F Boston Post Road, Sudbury, MA 01776)

Background sound is generally recognized as an important consideration in achieving acoustical privacy for open plan as well as enclosed

offices. Too often in the design of multifamily structures little systematic attention is given to the background sound levels that will actually exist in the completed dwelling spaces and therefore the ultimate acoustical privacy that will be experienced by the occupants. A number of recent case histories have been examined in terms of the sound isolation rating (NIC or STC) and background sound level (dBA) versus satisfaction. This effort suggests criteria in terms of both STC and background sound levels are necessary to adequately predict occupant satisfaction.

THURSDAY MORNING, 25 MAY 1989

REGENCY B, 8:30 A.M. TO 12:10 P.M.

Session PP. Structural Acoustics and Vibration V: Experimental Techniques in Structural Acoustics

Joseph M. Cuschieri, Chairman

Department of Ocean Engineering, Florida Atlantic University, Boca Raton, Florida 33431

Chairman's Introduction—8:30

Invited Papers

8:35

PP1. Scanning laser vibrometer system for structural intensity measurements. Alan D. Stuart, Martin J. Pechersky, Sabih I. Hayek, and B. C. T. Suen (Graduate Program in Acoustics and the Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802)

The steady-state real-time velocity response at successive locations on a beam are measured with a laser vibrometer and stored in a computer. The scanning system associated with the vibrometer consists of precision linear translation stages to control a lens/mirror combination. The laser beam is scanned along the length of the beam and the flexural vibration of a sinusoidally driven beam is measured at several locations. The amplitude and phase data are recorded at each location as well as a signal proportional to and in phase with the driving force. Knowing the distance between measurement locations, the recorded data are used to compute any of the following: spatial derivatives, transfer functions, and intensity.

9:00

PP2. Measurement of structural intensity in submerged plates and shells using nearfield acoustical holography. Earl G. Williams and Anthony J. Romano (Naval Research Laboratory, Code 5137, Washington, DC 20375-5000)

Nearfield acoustical holography provides a complete mapping of the normal surface velocity distribution on the surface of a plate or cylindrical shell. For plates this measurement completely determines the full structural intensity (SI) in the plate and no assumptions need to be made about the wave field on the plate. In shells this normal velocity is not sufficient to determine the SI accurately (below the ring resonance) because of the coupling between in-plane velocity and out-of-plane velocity. Several techniques are presented to predict the in-plane velocities from a knowledge of the normal (radial) velocity taken from measured holographic data and the corresponding SI is calculated and compared. In addition, the instantaneous energy exchange between the shell and the nearfield in the fluid (the coupling between structural and acoustic intensity) is obtained from measured data and presented.

PP3. The use of optical techniques for monitoring fluid-structure interactions. Yves H. Berthelot, Dowon Lee, and Jacek Jarzynski (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

The use of coherent light from laser sources has revolutionized the capabilities of measurement techniques in experimental fluid mechanics. In particular, laser Doppler velocimetry has become a standard tool for measuring nonintrusively the small-scale structure of turbulence in fluids. Similarly, laser vibrometers are becoming common instruments for detecting vibrations on solid surfaces. Interferometric measurements (homodyne or heterodyne) can provide very accurate surface vibration measurements, with resolution of a fraction of an Angstrom; subsequently, they are a powerful tool for monitoring fluid-structure interactions, especially for small scale phenomena. At larger scales, another powerful tool is the laser speckle and holographic interferometry which, when combined with optical Fourier transform techniques, can yield a direct visualization of wave propagation of the ω - k space (wave vector filtering). A recent effort has also been made to design a simple and rugged fiber optic laser probe that would be used to measure noninvasively the vector displacement of a vibrating structure. Such a probe allows the identification of both in-plane (short wavelength, acoustically inefficient) and out-of-plane (long wavelength, acoustically efficient) modes of vibration, and the conversion between these modes at discontinuities such as ribs on the surface. The relation between such vector measurements with acoustical holography and structural intensity will be discussed together with the use of scanning techniques. [Work supported by ONR.]

PP4. Structural intensity measurement techniques for single and multiple sources. Peter R. Wagstaff, Boudjema Bouizem, and Jean-Claude Henrio (Department Génie Mécanique, Université de Compiègne, B. P. 649, 60206 Compiègne, France)

The information available for power flow measurements may be used to determine the transmission paths of vibration sources. In multiple source situations, direct measurements of structural intensity do not give information on the contribution of each source. Reference signals associated with each source may be used to condition the spectra obtained at the measurement point to provide an estimate of the power flow due to each input. This technique has already been successfully applied to acoustic measurements. Results are presented using the direct method and these "selective techniques" for measurements using two, three, and four accelerometers, and for single and multiple inputs. The structure used for these experiments was a lightly damped bar and the results, using different types of reference signal and various estimators for the source contributions, illustrate the problems and the advantages of such methods in structural intensity measurements.

PP5. Experimental measurement of structural power flow on an aircraft fuselage. J. M. Cuschieri (Center for Acoustics and Vibrations, Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

An experimental technique is used to measure the structural power flow through an aircraft fuselage with the excitation near the wing attachment location. Because of the large number of measurements required to analyze the whole of an aircraft fuselage, it is necessary that a balance is achieved between the number of measurement transducers, the mounting of these transducers, and the accuracy of the measurements. Using four transducers mounted on a bakelite platform, the structural intensity vectors at locations distributed throughout the fuselage are measured. To minimize the errors associated with using a four-transducer technique the measurement positions are selected away from bulkheads and stiffeners. Because four separate transducers are used, with each transducer having its own drive amplifier, phase errors are introduced in the measurements that can be much greater than the phase differences associated with the measurements. To minimize these phase errors two sets of measurements are taken for each position with the orientation of the transducers rotated by 180° and an average taken between the two sets. In this paper results of the measurements will be presented and discussed. [Work supported by NASA Langley.]

Contributed Papers

PP6. The use of the Wigner distribution to identify structure-borne noise components. J. S. Bolton and T. J. Wahl (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

The Wigner distribution (W.D.) may be used to examine the time-frequency structure of a time history: e.g., to give an indication of signal arrival time *versus* frequency. The W.D. has recently found application in the study of wave scattering and in speech analysis. In this paper the use of the W.D. in structural acoustic problems will be illustrated. In a typical structure, vibrational energy is conveyed by several wave types simulta-

neously: e.g., flexural, torsional, longitudinal. Each wave type is characterized by a dispersion relation that may be interpreted in terms of phase speed as a function of frequency. In this paper it will be shown that the W.D. may be applied to structural impulse responses to reveal the dispersion relations of the wave types carrying significant energy between two points on a structure. Insofar as the dispersion relations may be associated with particular wave types, the W.D. may be used as the basis of an experimental procedure to identify the wave types present in a structure. Knowledge of this type is usually required to design optimum noise control treatments. Example calculations based on wave propagation in fluid-loaded plates and beams will be used to illustrate the proposed procedure.

10:55

PP7. Acoustic characteristics of confined jets. Part 1: Theory and numerical calculations. Kam W. Ng (Naval Underwater Systems Center, Newport, RI 02841-5047) and Peter R. Stepanishen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881)

A theoretical model was developed to study the wall pressure and vibratory motion of an elastic pipe which is excited by a confined turbulent jet flow resulting from fluid flow through orifices in the pipe. Based on flow field measurements, the blocked surface pressure was calculated using Lighthill's method, and then used to drive the fluid-filled shell. The wall pressure and pipe wall acceleration were determined by solving the coupled fluid-solid interaction problem. The wall pressure was obtained by summing the blocked surface pressure and the pressure due to the wall vibration. An amplitude-modulated convecting wave field was used to simulate the moving acoustic sources of the jet. The random nature of the turbulent jet was incorporated into the analytical model. Specifically, the acoustic pressure was assumed to result from hydrodynamic pressure fluctuations which are uncorrelated in the radial direction, but correlated in the axial direction near the jet exit. The uncorrelated pressure fluctuations in the radial direction reflect the random motion of the turbulent jet, whereas the correlated pressure fluctuations in the axial direction reflect the motion of the large-scale coherent structures near the jet exit. It has been demonstrated that the noise model is capable of relating the flow field and the acoustic field of confined jet flows. [Work supported by ONT and ONR.]

11:10

PP8. Acoustic characteristics of confined jets. Part 2: Measurements. Kam W. Ng (Naval Underwater Systems Center, Newport, RI 02841-5047) and Peter R. Stepanishen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881)

The flow and acoustic characteristics of confined jets were measured by a laser Doppler velocimeter, flush-mounted hydrophones, and accelerometers. Unrestricted pipe flow and flows restricted by various orifices were tested for a wide range of velocities to simulate the flow in piping systems. Wall pressure data showed that the noise levels vary with the pipe's axial location and the peak noise is located in the vicinity of the end of the jet potential core. Correlations of wall pressure fluctuations with jet velocity showed a velocity to the 3.7th to 5.0th power relationship. A nondimensional wall pressure spectrum was established for the various confined jets by the Strouhal relationship, where the length scale is the jet hydraulic diameter. This jet pressure spectrum agrees well with the wall pressure spectrum of a turbulent boundary layer above a rigid plane. Correlations of wall pressure fluctuations and pipe wall acceleration signals showed that jet flows generate more deterministic features than pipe flow. The coherence functions of the wall pressure and pipe wall acceleration signals are relatively high near the exit of the jet. The high coherence is probably due to the large-scale coherent structures. Experimental results for wall pressure and acceleration compared reasonably well with the theoretical model for a wide range of parameters of interest. [Work supported by ONT and ONR.]

11:25

PP9. Resonant excitation of interface waves on a compliant coating in the presence of a fluid flow. Maryline Talmant, Xiao-Ling Bao, H. Überall (Department of Physics, Catholic University of America, Washington, DC 20064), and W. Madigosky (Naval Surface Warfare Center, White Oak, Silver Spring, MD 20903)

The stability of interface waves on a compliant coating, attached to a rigid substrate and exposed to a laminar or turbulent fluid flow, has been the subject of several theoretical studies, and of experimental investigations in which the onset and growth of instability waves was determined. Solutions of the characteristic equation of the problem have been obtained, which furnish dispersion curves and stability information for the interface waves, assuming a compressible fluid in order to attain the correct static limit. It has been shown that the excitation of these interface waves can occur in a resonant fashion; using this approach, an accurate experimental determination of the dispersion curves, the onset of instability, and the growth properties of interface waves will be possible.

11:40

PP10. Modeling of transient vibration envelopes using statistical energy analysis techniques. Ming-Lai Lai and Andres Soom (Department of Mechanical and Aerospace Engineering, University at Buffalo, Buffalo, NY 14260)

The modeling, by the statistical energy analysis (SEA) method, of transient vibration envelopes of coupled systems is investigated. The relation between the time-varying energy transferred between two coupled subsystems and time-varying energies of the individual subsystem is studied numerically and experimentally. These studies indicate that time-varying energy transmitted between two subsystems can be related to the subsystem energies by an apparent time-varying coupling loss factor. It is shown that the apparent coupling loss factor approaches the asymptotic (or steady-state) coupling loss factor as response energies and transferred energies are integrated over progressively larger times. Both the apparent time-varying coupling loss factor and the asymptotic coupling loss factor, determined experimentally, are used in energy balance equations to predict the time-varying vibration envelopes of a system of two point-coupled plates and the results are compared. Although overall response predictions are similar, considerable differences are noted in individual frequency bands.

11:55

PP11. A nonlinear theory for high-frequency vibrations of piezoelectric crystal rods. M. Cengiz Dökmeci (Faculty of Aeronautics and Astronautics, Istanbul Technical University, P.K. 9, Istanbul 80191, Turkey)

This paper is addressed to a systematic derivation of a hierarchy of one-dimensional theories for the motions of crystal rods in view of the author's review article [Shock Vib. Dig. 20(2), 3-20 (1988)]. The nonlinear theories are derived using a unified variational principle [IEEE UFFC, 35(6) (November 1988)] together with the trigonometric series expansions of the field quantities [cf. Int. J. Solids Structures 10(4), 401-409 (1974)]. All the mechanical and electrical effects and the material nonlinearity are taken into account. The resulting equations govern all the extensional, thickness, flexural, torsional, and coupled motions of crystal rods. By a proper truncation of the series expansions, special motions and material of crystal rods are considered. Also, the fully linearized equations of crystal rods are examined. The sufficient boundary and initial conditions are enumerated so as to ensure the uniqueness in solutions of the linearized rod equations. [Work supported by U.S. Army through its European Research Office.]

Meeting of Accredited Standards Committee S12 on Noise

to be held jointly with the

Technical Advisory Group for ISO/TC 43/SC1 Noise and ISO/TC 94/SC12 Hearing Protectors

W. Melnick, Chairman S12

Ohio State University, University Hospital Clinic, 456 W. 10th Avenue, Columbus, Ohio 43210

H. E. von Gierke, Chairman, Technical Advisory Group for ISO/TC 43/SC 1

1325 Meadow Lane, Yellow Springs, Ohio 45387

Standards Committee S12 on Noise. Working group chairs will report on their progress under the plan for the production of noise standards. The interaction with ISO/TC 43/SC1 and ISO/TC 94/SC12 activities will also be discussed, with reference to the international standards under preparation. The Chair of the respective U.S. Technical Advisory Groups (H. E. von Gierke) will report on current activities of these committees.

THURSDAY MORNING, 25 MAY 1989

REGENCY C, 8:45 A.M. TO 12:05 P.M.

Session QQ. Underwater Acoustics VI: Spatial and Temporal Variability of Ambient Noise I

Frederick H. Fisher, Chairman

Marine Physical Laboratory, P-001, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, California 92093

Chairman's Introduction—8:45

Invited Papers

8:50

QQ1. Remote sensing of the ocean bottom using the vertical directionality of the ambient noise in the water column. Michael J. Buckingham (Mission Management Department, Royal Aerospace Establishment, Farnborough, Hampshire GU14 6TD, England)

The vertical directionality of ambient noise in the shallow ocean overlying a fast, fluid sediment is a stable feature of the noise field. It is controlled by the reflection properties of the bottom, most notably the fact that the water/sediment interface shows a critical grazing angle α_c . Noise rays that are incident on the bottom at grazing angles greater than α_c penetrate the interface and are lost to the water column, whereas rays with shallower grazing angles are totally internally reflected. The latter remain trapped in the channel, and contribute to the noise field. Thus, the noise field shows a symmetrical peak in the horizontal whose angular width is $2\alpha_c$. Measurements of the vertical directionality of the ambient noise have been made at six sites around the U.K. coast using air-deployed vertical line arrays. From these data, estimates of the critical grazing angle have been obtained. Since the sound speed in the water column was also known from AXBT data, it was possible to infer from the measured values of α_c the sound speed in the sediment. A comparison of the sediment sound speed obtained from the new noise technique with the results obtained by an independent survey using conventional seismic methods shows excellent agreement.

QQ2. Simulation of array processing in three-dimensional noise fields: Suggested experiments. John S. Perkins, Bruce H. Pasewark, and W. A. Kuperman (U.S. Naval Research Laboratory, Washington, DC 20375-5000)

The growing capability to model three-dimensional noise fields in complex ocean environments taken together with concurrent recent developments in matched-field processing (MFP) suggests noise experiments that will shed light onto the physical processes generating and propagating the noise fields. These experiments should especially yield information concerning the noise field as it impacts on signal and/or array processing. Some of the expected (modeled) characteristics of low-frequency ambient noise in a deep ocean basin with realistic complexity and its subsequent impact on MFP are reviewed. For example, the effects on MFP of down-slope conversion from discrete sources (surface ships) and distributed sources (storms), and the effectiveness of various rejection techniques are demonstrated. These results suggest experiments to investigate the most important processes.

QQ3. Low-frequency ocean ambient noise variability: Measurements. David G. Browning, William M. Carey, and William A. Von Winkle (Naval Underwater Systems Center, New London, CT 06320)

The interaction of the wind with the ocean surface is recognized as a major source of ambient noise, however, at low frequencies (< 200 Hz) it is often difficult to isolate wind-generated noise from shipping noise. This is further complicated by a relatively low transmission loss at these frequencies which allows possible noise contributions from long distances and various propagation paths. An analysis is made of historic ambient noise data based on recent developments in noise generation and sound propagation theory to sort out the ship-generated noise component. The results appear to support two distinct wind-generated noise mechanisms which may be delineated, as suggested by Kerman, by a critical surface velocity.

QQ4. Ice edge noise "hot spots." Orest Diachok, Patricia Gruber, Peter Mitchell (Naval Research Laboratory, Code 5120, Washington, DC 20375), and George Vermillion (U.S. Naval Station, Rota, Spain)

Previously reported and new measurements of the noise field at the periphery of the Arctic Ocean will be reviewed. Previous measurements with sonobuoys reveal that ambient noise levels are usually high at the ice edge, and that levels increase with wind speed, ice concentration, and fetch. Subsequent measurements of the noise with a towed array (by Yang *et al.*) indicate that the edge noise is concentrated at spatially localized (< 5 km), widely separated (nominally 30 km) "hot spots." To shed more light on this noise source and its predictability, an experiment was conducted with a towed array to probe the hot spots acoustically with a ship-towed array, and electromagnetically with an aircraft-mounted passive microwave imager, a ship-mounted radar, and satellite sensors. Results of these measurements and their implications for the predictability of hot spot locations, effective source levels, and long-range effects will be presented.

Contributed Papers

QQ5. Ice edge ambient noise. Ola M. Johannessen, Hanne Sagen (Nansen Remote Sensing Center, Edv. Griegsv. 3A, N-5037 Solheimsvik, Norway), Kenneth V. Starke (Naval Ocean System Center, San Diego, CA 92152), Ingjald Engelsen (Norwegian Defence Res. Establishment, Horten, Norway), and Susan Payne (Applied Research Laboratory, University of Texas, Austin, TX 78712)

Ambient noise data along with oceanographic, meteorological, and remote sensing data were collected during the Marginal Ice Zone Experiment in March and April 1987 in the Greenland Sea. The experiment was designed to study how ambient noise was generated by ice kinematics, ice edge eddies, and wave propagation into the ice pack. Sonobuoys were deployed in the water off the ice edge and in the adjacent ice-field where eddies, jets, and surface waves were present. SAR images of the area were obtained during the ambient noise recordings and provided important information such as position of the ice edge relative to the sonobuoys. Advanced image processing enabled ice classification, and estimates of ice

concentration and floe size distribution. These parameters provided information about how ice can influence the ambient noise level and frequency distribution. Procedures to enhance wave signature in the SAR images were used to analyze the wave pattern. Correlation between SAR and ambient noise data support the hypothesis that eddies and jets represent noise level anomalies along the ice edge.

QQ6. Measurements of ambient noise "hot spots" along the ice edge in the Greenland Sea Marginal Ice Zone. Kristian P. Biggs (Naval Postgraduate School, Monterey, CA 93943), Patricia L. Gruber (AT&T Bell Laboratories, 1201 S. Hayes Street, Arlington, VA 22202), and Robert H. Bourke (Naval Postgraduate School, Monterey, CA 93943)

During July 1987, an experiment was conducted to measure ambient noise along the ice edge in the Greenland Sea. Ambient noise "hot spots," concentrated areas of relatively high noise levels, were located along the ice edge using a towed array. Ambient noise levels were then recorded using sonobuoys deployed from P3 aircraft. The sonobuoy data were averaged in time to obtain the noise spectra below 2 kHz. The temperature structure of the area was determined by XBT and AXBT buoys and the ice edge location was determined from coincident satellite photos, 90-GHz microwave imagery, and P3 radar ice edge maps. The relationship between the environmental and acoustic data suggests the existence of hot spots which are (1) primarily related to wind and sea state and (2) primarily related to ice edge compaction by eddy flow.

11:20

QQ7. Update on an evolving model for predicting thermally induced noise from Arctic pack ice. Peter J. Stein (Atlantic Applied Research Corp., 4 "A" Street, Burlington, MA 01803), James K. Lewis (Science Applications International Corporation, 1304 Deacon, College Station, TX 77840), and Warren W. Denner (Science Applications International Corporation, 205 Montecito Avenue, Monterey, CA 93940)

Fracturing of Arctic pack ice due to thermal stresses is known to be a source of higher frequency (> 300 -Hz) Arctic ambient noise. Recently, Lewis and Denner [J. Acoust. Soc. Am. **84**, 1444 (1988)] reported on a numerical model for predicting temperature profiles through the ice. They showed that ice temperatures, and thus the stresses that lead to fracturing, are a function of sensible heat flux, solar radiation, and long wave radiational heat flux. This implies a dependence on snow and cloud cover and explains some of the variations in higher frequency noise. Here an extension of the model to include estimation of stresses in the ice via a stress-strain rheology that includes creep is reported. These calculations can be used to predict failure of the ice at a given depth in the ice. The ice was found to fail not only near the surface, where the maximum temperature variations occur, but also deep within the ice. Acoustic data show that the radiation from the surface fracturing is stronger than the radiation from the deep fracturing. Variations in acoustic source strength for a crack in the ice as a function of depth within the ice may explain this feature. [Work supported by ONR.]

11:35

QQ8. A switched-source model for simulating Arctic undersea noise. Samuel M. Nagle (Honeywell Marine Systems Division, M/S 3D02, 6500 Harbour Heights Parkway, Everett, WA 98204-8899)

A switched-source model is developed for simulating the impulsive character of Arctic undersea noise. The model is based on one proposed by S. V. Czarnecki ["Nearly optimal detection of signals in non-Gaussian noise," Ph.D. thesis, Dept. of EECS, Princeton Univ. (October 1983)] where impulsive and normal background noise regions are treated as coming from two independent Gaussian sources with differing variance. P. K. Willet and J. B. Thomas [IEEE J. Ocean. Eng. **OE-12**, 29-37 (1987)] have suggested that the dual mode noise sequence can be treated as stationary and dependent if the state of the source selection switch is governed by a two-state Markov process. In this paper a statistical analysis is conducted on a 10-min time series of Fram 2 noise data in order to provide a complete stochastic description of real noise data within the framework of such a model. An earlier point process analysis is extended by examining the duration and amplitude of burst events. Results of the analysis are used to specify the dual source characteristics, and the terms of the transition matrix for the Markov process.

11:50

QQ9. Simulating and processing the Arctic acoustics due to ice dynamics. C. A. Pomalaza-Raez, S. S. Shanan (Department of Electrical and Computer Engineering, Clarkson University, Potsdam, NY 13676), and H. H. Shen (Department of Civil and Environmental Engineering, Clarkson University, Potsdam, NY 13676)

Under sea-ice noise, in particular that produced by ice floes in the Arctic region, is studied by means of a model simulation and the development of signal processing techniques for its measurement and estimation. This noise is due to the interaction among ice floes that produces non-Gaussian noise in time and frequency. Based on previous studies, a Gaussian shape acoustic wave pressure generated by a single collision is assumed. The random occurrence of the ice collision process is modeled as a Poisson sequence of impulses. Different sonar array spectrum estimation techniques [Fourier, autoregressive (AR), and minimum variance (MV)] are analyzed for the processing of the resulting acoustic signal. It is found that the minimum variance spectral estimation technique is the overall best for this type of signal for a wide range of the model parameters. Previously reported studies have often used the periodogram technique that as shown here usually gives poor results.

THURSDAY MORNING, 25 MAY 1989

NEWHOUSE I, ROOM A1, 9:45 TO 11:30 A.M.

Session RR. Psychological Acoustics VI: General Psychoacoustics

Richard Freyman, Chairman

Department of Communication Disorders, University of Massachusetts, Amherst, Massachusetts 01003

Contributed Papers

9:45

RR1. Computer simulations of mutable psychometric functions. Marjorie R. Leek (Army Audiology & Speech Center, Walter Reed Army Medical Center, Washington, DC 20307-5001), Thomas E. Hanna, and Lynne Marshall (NSMRL, Box 900, SUBASE NLON, Groton, CT 06349)

The psychometric function of an experimental subject is almost certainly not fixed, but varies in response to nonsensory factors such as attention and learning. This is particularly true when the subject is inexperienced, or the complexity of the task requires a substantial allocation of attentional resources by the subject. With commonly used psychometric procedures it is impossible to know whether a shallow psychometric function slope is a true reflection of the sensory process, or is a result of "aver-

aging" a highly variable underlying function. Computer simulations were used to assess the ability of a 2IFC adaptive procedure with two interleaved tracks to monitor the stability of the underlying psychometric function over the course of an experiment. Variability due to the slope of the underlying function and the measurement procedure will be reflected in the average level difference between the two tracks (across-track measurement), while the additional variability due to a changing psychometric function will be reflected in the within-track level differences. Simulated psychometric functions were either fixed across all trials, shifted along the "stimulus" axis according to either a sinusoidal or square-wave movement (to simulate drifts or lapses of attention, respectively), or shifted by a negative exponential function to simulate learning effects. The across-track variability remained nearly constant across all attentional conditions, and was dependent almost entirely on the underlying slope of the function. However, within-track variability was strongly affected by the changes in psychometric function location. An estimate of underlying slope may be obtained from the across-track variability, and an indication of the relative stability of the underlying function can be extracted from within-track variability. [Work supported by the VA.]

10:00

RR2. Effect of threshold on the growth of loudness at low frequencies. Walt Jesteadt (Boys Town National Institute, 555 N. 30th Street, Omaha, NE 68131) and Larry E. Humes (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

Loudness is generally thought to increase as a function of intensity at a more rapid rate at frequencies below 500 Hz than at higher frequencies. It is well established that loudness increases at a more rapid rate near threshold for all frequencies. To what extent can the pattern of loudness growth at low frequencies be accounted for by a combination of the threshold effect and the marked increase in threshold below 500 Hz? To answer this question, equations incorporating the threshold effect were fitted to several sets of data in the literature describing growth of loudness at low frequencies. Two such equations are generally used: $\psi = k(\phi - \phi_0)^\beta$ or $\psi = k(\phi^\beta - \phi_0^\beta)$, where ψ represents the psychological magnitude, ϕ represents the stimulus intensity, and ϕ_0 represents stimulus intensity at threshold. With the exponent β held constant across frequencies and k allowed to vary, either equation can account for very high percentages of variance in the data and the residual error variance is not systematic as a function of frequency. This analysis suggests that the exponent of the loudness function does not change with frequency. [Work supported by NIH.]

10:15

RR3. The slope of the loudness function for individuals and groups with normal and impaired hearing. Rhona P. Hellman and Carol H. Meiselman (Auditory Perception Laboratory, Northeastern University, Boston, MA 02115)

The slope of the loudness function for pure tones is derived at a frequency in the region of impaired hearing for more than 90 people with bilateral moderate-to-severe cochlear hearing losses. Calculations of the slopes are obtained over the stimulus range where cochlear impairment steepens the loudness function. Slope distributions for impaired hearing, determined from magnitude estimation and production and from cross-modality matching, are compared to those obtained in a similar manner for 51 people with normal hearing. Values of the slopes reveal two distinct populations. Most of the slopes in normal hearing vary from 0.4 to 0.7 with a mean close to 0.6; in cochlear impairment they vary from 1.1 to 2.8 with a mean close to 2.0. In accord with loudness matching, much of the greater variability of the impaired distributions arises from the heterogeneity of thresholds. The results also show that group loudness functions generated from cross-modality matching are consistent with theoretical functions determined from an equation for loudness suggested by Zwillocki [Handbook Math. Psychol. 3, 3-97 (1965)]. [Work supported by the Rehabilitation Research and Development Service of the VA.]

10:30

RR4. Perceived distance and loudness-softness functions. Miguelina Guirao (Auditory Perception Laboratory, Northeastern University, Boston, MA 02115 and Laboratorio de Investigaciones Sensoriales CONICET, Facultad de Medicina UBA, Buenos Aires, Argentina)

Experiments were designed to investigate the relation between auditory-perceived distance and sound intensity. A pure tone at 1000 Hz and a wide band of noise were heard through earphones in a soundproof room. Sounds were presented at random and at continuously increasing or decreasing levels over a stimulus range from 20 to 80 dB SPL. Naive listeners with normal hearing estimated distance by judging nearness (proximity) and farness (remoteness). They also judged loudness and softness. Judgments were made by a conventional and two modified versions of the method of magnitude estimation. According to preliminary results, the effect of stimulus level on the responses depends on whether the subjects are instructed to judge the sound in terms of its nearness to or distance from them. The nearness and farness functions are similar in shape to the loudness and softness functions, respectively. Results are discussed in relation to the auditory attributes of density and volume. [Work supported by NIH.]

10:45

RR5. Pure-tone intensity discrimination in hearing-impaired and normal-hearing listeners. Anna C. Schroder, David A. Nelson (Department of Otolaryngology, University of Minnesota, Minneapolis, MN 55455), and Neal E. Viemeister (Department of Psychology, University of Minnesota, Minneapolis, MN 55455)

Intensity jnd's for 500-ms tones were obtained from five cochlear-impaired and three normal-hearing subjects. The jnd-level functions, from threshold to 97 dB SPL, were measured at five test frequencies, in quiet and in high-pass noise. The "near-miss" to Weber's law was observed from normals in quiet, from two subjects with flat losses, and from regions of normal threshold in three subjects with high-frequency loss. In the same three ears, at frequencies just below and within the region of loss, jnd's conformed to Weber's law, as did the jnd's obtained in the presence of the noise for all subjects. These results are consistent with explanations of the near-miss that involve excitation at frequency regions well above the test frequency. At equal SPLs, jnd's in quiet from impaired subjects were equal to, or larger than, jnd's from normals. At equal SLs, jnd's in quiet from hearing-loss subjects were sometimes smaller than normal, but were usually elevated by high-pass noise, thus implicating the near-miss. In some cases, however, jnd's at low SLs were still smaller in the noise, which may reflect the steeper rate-level functions seen in the auditory nerve in regions of cochlear loss [M. C. Liberman and N. Y.-S. Kiang, *Hear. Res.* 16, 75-90]. [Work supported by NINCDS Grant NS12125.]

11:00

RR6. Auditory filter shapes at low frequencies. Robert W. Peters (Speech and Hearing Sciences, Department of Medical Allied Health, University of North Carolina, Chapel Hill, NC 27514) and Brian C. J. Moore (Department of Experimental Psychology, University of Cambridge, Cambridge CB2 3EB, England)

Auditory filter shapes were measured in normally hearing subjects for center frequencies (f_c) of 100, 200, 400, and 800 Hz using the notched-noise method [R. D. Patterson and I. Nimmo-Smith, *J. Acoust. Soc. Am.* 67, 229-245 (1980)]. Two noise bands, each $0.4f_c$ wide, were used; they were placed both symmetrically and asymmetrically about the signal frequency, to allow the measurement of filter asymmetry. The overall noise level was always 80 dB SPL. Stimuli were delivered monaurally using Sennheiser HD424 earphones. Except for $f_c = 100$ Hz, the auditory filters were asymmetric, the upper skirt being steeper than the lower skirt. At 100 Hz the filters were more symmetric, or showed an asymmetry in the opposite direction. This can probably be attributed to the reduced effect-

tiveness of the lower noise band due to the transmission characteristics of the outer and middle ear. The equivalent rectangular bandwidths of the filters had average values of approximately 29, 42, 74, and 160 Hz for values of f_c of 100, 200, 400, and 800 Hz, respectively. The signal-to-masker ratio at the output of the filter required to achieve threshold tended to increase with decreasing f_c .

11:15

RR7. On the pitch of mistuned harmonics in a complex tone. William Morris Hartmann and Sandra L. Smith (Department of Physics, Michigan State University, East Lansing, MI 48824)

Four listeners matched the pitch of a single mistuned harmonic in an otherwise periodic complex tone, whose fundamental frequency was near

200 Hz. In individual experiments, harmonics numbers 1, 2, 3, 4, 5, 7, 9, and 11 were individually mistuned, by 11 different amounts ranging from -8% to $+8\%$. The experiment thus measured the shift in the pitch of a sine component caused by other components of the complex tone. The experiments showed that for mistuned low-order harmonics a positive mistuning produces a positive pitch shift while a negative mistuning produces a negative pitch shift. For mistuned high-order harmonics the pattern is similar except that the pitch shifts caused by positive mistunings become large. These data were compared with a place theory in which the peak in the excitation pattern for an individual harmonic is partially masked by peaks due to neighboring harmonics. Although the model correctly predicts the results for positive mistunings, it fails dramatically for negative mistunings. The data are also compared with a theory whereby pitch is determined by peaks in an interspike-interval histogram *per* the MIT model. This theory can account for all the data, except for the mistuned fundamental, though filter bandwidths must be considerably enlarged compared to the MIT model to obtain agreement. [Work supported by the National Institutes of Health.]

Session SS. Architectural Acoustics IV and Musical Acoustics VI: Music Education Facilities Since 1975 (Poster Session)

Edward R. McCue, Chairman
Wenger Corporation, 555 Park Drive, Owatonna, Minnesota 55060

Contributed Papers

All posters will be on display from 1:00 to 5:00 p.m. (Viewing hours will also be extended into the evening.) To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 to 5:00 p.m. Contributors are encouraged to leave their posters in place until 9:45 p.m. A cash bar will be set up in the evening to facilitate informal discussion. The Goldstein Auditorium will be closed at 10:00 p.m.

Posters will be presented by the following persons and organizations:

David L. Adams Associates, Inc.	Klepper Marshall King Associates, Ltd.
Artec Consultants	Marshall Long Associates
Bolt Beranek and Newman Inc.	NHK Engineering Services, Inc. (Japan)
Campanella Associates	Minoru Nagata Acoustic Engineer & Associates Co., Ltd. (Japan)
Cavanaugh Tocci Associates, Inc.	Paoletti/Lewitz/Associates, Inc.
Coffeen Fricke & Associates, Inc.	Adviesbureau Peutz & Associes B. V. (The Netherlands)
Office of M. David Egan, Consultants in Acoustics	Gerald A. B. Riley (Australia)
Harold Geerdes Consulting Services	TNO Institute of Applied Physics (The Netherlands)
Peter George Associates, Inc.	Peter Terroux, Consultant in Acoustics (Canada)
Jaffe Acoustics, Inc.	Towne, Richards & Chaudiere, Inc.
Adrian W. D. Jongens (South Africa)	Wilson, Ihrig & Associates, Inc.
R. Lawrence Kirkegaard and Associates	

Session TT. Education in Acoustics I: Demonstrations of Experiments in Acoustics and Undergraduate Posters (Lecture and Poster Session)

Mark F. Hamilton, Cochairman

Department of Mechanical Engineering, University of Texas at Austin, Austin, Texas 78712

Murray S. Korman, Cochairman

Department of Physics, U. S. Naval Academy, Annapolis, Maryland 21402

Chairman's Introduction—1:00

Invited Papers

1:05

TT1. Demonstrations of underwater noise generation by bubbles. Lawrence A. Crum and Hugh C. Pumphrey (National Center for Physical Acoustics, University of Mississippi, University, MS 38677)

In 1933, Minnaert demonstrated that the sound of a babbling brook was due to the oscillations of gas bubbles entrained in the water [M. Minnaert, *Philos. Mag.* **16**, 235 (1933)]. In 1959, Franz discovered that rain drops could produce relatively large amounts of underwater noise through the entrainment of gas bubbles during drop impact [G. J. Franz, *J. Acoust. Soc. Am.* **31**, 1080 (1959)]. In 1988, Prosperetti suggested that a major contribution to ocean ambient noise was due to bubbles [A. Prosperetti, *J. Acoust. Soc. Am.* **84**, 1042 (1988)]. A series of demonstration experiments will be presented that are not difficult to assemble and that will show that gas bubbles are a major source of underwater noise for a variety of hydrodynamic situations. [The research leading to the understanding of these effects has been supported by the ONR.]

1:30

TT2. Demonstrations of nonlinear oscillators, parametric excitation, and solitons. Robert Keolian (Applied and Engineering Physics, Cornell University, Ithaca, NY 14853)

The peculiar behavior of nonlinear oscillations will be demonstrated with a series of experiments selected from those shown previously in Austin [*J. Acoust. Soc. Am. Suppl.* **1 77**, S35 (1985)]: (1) A stretched rubber band, driven by a loudspeaker, exhibits hysteresis due to its bent resonance curve. (2) A doubly bent resonance curve and hysteresis are exhibited by a parametrically driven pendulum bouncing against stops. (3) A hanging chain undergoes quasiperiodically modulated vibrations, where the modulations have a frequency independent of that of the drive. (4) A parametrically driven rigid pendulum will defy gravity by standing on end and oscillating upside down. (5) Standing waves on a trough of water will be parametrically driven, as seen by Michael Faraday. (6) A nonpropagating soliton [*Phys. Rev. Lett.* **52**, 1421 (1984)] will be shown in a trough of water, and two of these solitons will oscillate about each other.

1:55

TT3. Education, research, and acoustic levity: Miscellaneous demonstrations. J. D. Maynard (The Pennsylvania State University, 104 Davey Lab, University Park, PA 16802)

During the course of teaching and conducting research in acoustics, a number of demonstrations have been developed. Some have been for classroom education, some have been for our own education, some have been for entertainment, and some have led to serious research projects. These demonstrations include the sonic motor, acoustic levitation, the projected Chladni plate, shock inception, noise riding sound, following a mass and spring to chaos, acoustic Anderson localization, and tuning up a quasicrystal. The demonstrations will be presented as permitted by time and transportation limitations.

2:20

TT4. A simple apparatus for the acoustic levitation of a drop in a liquid. Robert E. Apfel (Department of Mechanical Engineering, Yale University, P. O. Box 2159, New Haven, CT 06520)

A simple apparatus consisting of a piezoelectric disc, beaker, function generator, audio amplifier, current probe, and oscilloscope is sufficient to demonstrate the acoustic levitation of a liquid drop in an immiscible host liquid. Applications of acoustic levitation, which include measuring the tensile strength of liquids, sound velocity, interfacial tension, and other properties, will be discussed. [This has been supported by the Office of Naval Research.]

2:55

TT5. Simple demonstration of a thermoacoustic sound source. Thomas J. Hofer, Anthony A. Atchley, and Steven L. Garrett (Physics Department, Code 61, Naval Postgraduate School, Monterey, CA 93943)

A simple thermoacoustic prime mover, known as the "Hofer tube," consisting of an open-closed tube containing many G-10 fiberglass plates will be demonstrated. Sound of sufficiently large amplitude to produce streaming is generated when a temperature gradient is imposed across the thermally insulating plates. The use of helium gas rather than air as the thermodynamic working fluid provides a dramatic demonstration of the effects that sound speed and other thermal properties, such as the polytropic coefficient, have on the efficiency of acoustic power generation from the thermoacoustic process. A simple model will be provided to explain the engine behavior to a nonscientific audience.

3:20

TT6. Demonstration of computer-based sonograph display. Alan D. Stuart and Glenn Frey (Graduate Program in Acoustics, The Pennsylvania State University, University Park, PA 16802)

A computer-based sonograph display has been developed that shows the amplitude of the frequency spectrum as a function of time. Depending on the speed and processing capabilities of the computer, real-time displays can be obtained. The data acquisition procedure developed along with the display provides a means for conducting both detailed statistical analysis of the data as well as subjecting the data to other post processing analysis.

Contributed Papers

3:45

TT7. Demonstrations on the scattering of ultrasound by turbulence in water. Edward C. Zurey and Murray S. Korman (Department of Physics, United States Naval Academy, Annapolis, MD 21402)

A beam of monochromatic sound incident on a turbulent velocity field (produced by a submerged water jet) is scattered by the velocity fluctuations in the flow. Angular measurements of the Doppler shift and spectral broadening in the received frequency spectrum of the scattered sound intensity help to characterize the mean flow and rms velocities of the turbulent field, respectively. Scattered spectra are displayed for comparison in the cases where the incident beam is generated by a point source, a plane circular array, and a focused spherical array transducer. Results are shown in an arrangement where the focused beam scans across the width of the jet. Scattering theories are briefly mentioned in light of spectral broadening results.

The sing-around velocimeter [M. Greenspan and C. Tschiegg, *Rev. Sci. Instrum.* **28**, 897-901 (1957)] is capable of great precision and is an elegant measurement technique. It is also relatively easy to implement with low-cost electronics. Besides the obvious experiments on the dependence of sound speed in gases on temperature and molecular weight, it is also possible to measure gas diffusion rates or demonstrate the adiabatic nature of acoustic oscillations. Absolute measurement of sound speed within 1% is easily done. These demonstrations were developed for a graduate-level thermodynamics course given at the Pennsylvania State University Great Valley Graduate Center.

4:00

TT8. Measurement of the sound speed in air by sing-around velocimetry. Thomas B. Gabrielson (1216 Barness Drive, Warminster, PA 18974)

4:15

TT9. Videotape demonstrations of auditorium design principles using acoustic scale models. Gary W. Siebein (Department of Architecture, 231 ARCH, University of Florida, Gainesville, FL 32611)

A videotape demonstration was developed to illustrate acoustical design principles for auditoria to undergraduate architecture students. An

acoustical scale model of an auditorium was constructed with removable walls, floor, and ceiling. Starting with a microphone in free space, an enclosure is built in steps around the source and receiver. An echogram is recorded at each step so students can see the effect of increased enclosure on the sound that arrives at the listener's position. Theater seating, various quantities of absorbent material, acoustically shaped ceiling panels, and diffusing elements are added to the enclosure as the demonstration progresses. Echoes, excessive loudness, flutter, and other basic acoustical problems and their solutions are presented. The demonstrations show changes in the impulse response of the room as changes are made in the shape and volume of the room. It serves as an introduction to acoustical modeling and auditorium design for the students.

TT10. Sources for demonstration experiments in acoustics. Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

Many teachers, since the time of Chladni, Helmholtz, and Rayleigh, have used demonstration experiments to enhance their students' understanding of acoustical phenomena. Some sources for descriptions of suitable experiments for lecture demonstrations, corridor demonstrations, and laboratory use are discussed, as well as the author's experience with using demonstrations experiments in acoustics courses at various levels of instruction.

Poster Papers

Contributors will be at their posters from 4:30 to 5:30 p.m.

TT11. Simultaneous SLAM imaging and tensile loading of copper films. Wendy C. Crone (Department of Theoretical and Applied Mechanics, University of Illinois, 216 Talbot Lab, 104 S. Wright Street, Urbana, IL 61801)

Preliminary experiments on imaging the mechanical properties of copper films (1 and 5 mil thick) have been performed with the scanning laser acoustic microscope (SLAM). Assessment of material properties can be obtained through attenuation and speed measurements with the SLAM. Transmission curves for a copper film were obtained at 24 and 30 MHz to assess the effects of thickness and stiffness of the film. Predictions of optimum insonification angle were then determined for 100 MHz. Real time images of a copper film with a small crack were obtained while stretching was induced. Results are encouraging and the images display changes in the attenuation and speed in the region of the crack tip as the film is loaded. It is suspected that these changes are caused by the plastic deformation near the crack tip. Further experiments are being conducted to verify this and to measure other mechanical properties. [Advisor: John Harris. Work done in cooperation with SONOSCAN, Inc. and the Bio-acoustics Research Laboratory of the University of Illinois.]

TT12. Focusing of an N wave by an ellipsoidal reflector: An experiment in air. Stephen T. W. Cheng, Matthew R. Jones, David T. Blackstock, and

Wayne M. Wright (Applied Research Laboratories, The University of Texas at Austin, P. O. Box 8029, Austin, TX 78713-8029 and Mechanical Engineering Department, The University of Texas at Austin, Austin, TX 78713-1063)

An airborne experiment to model the intense pulse field developed by a lithotripter is in progress. The reflector is half an ellipsoid, machined from an aluminum block and having the following dimensions: major axis 280 mm, minor axis 140 mm, eccentricity 0.866, distance from aperture plane to either focus 121 mm. An electric spark at the interior focal point generates an N wave pulse. The reflected wave in the exterior region is observed with the aid of a condenser microphone having a very wide bandwidth. Of particular interest is the sequence of axial measurements from the aperture to the exterior focus and beyond. Expectations based on linear theory were that the pulse would start out N-shaped near the aperture, become U-shaped in the neighborhood of the focus (because of phase changes characteristic of three-dimensional focusing), and take on the shape of an inverted N beyond the focus. Measurements confirm expectations for the prefocal and focal regions. Beyond the focal region, however, the shape is not an inverted N. Instead it more closely resembles the U-shaped pulse recorded at the focus. [Work supported in part by Office of Naval Research and Texas Advanced Research Program.]

Session UU. Engineering Acoustics VII: Acoustic Measurements and General Topics**Robert D. Finch, Cochairman***Department of Mechanical Engineering, University of Houston, Houston, Texas 77204***Ilene J. Busch-Vishniac, Cochairman***Department of Mechanical Engineering, University of Texas, Austin, Texas 78712***Contributed Papers****1:00**

UU1. A method for measuring dynamic properties of soils. Jerry E. Farstad (Research Department, Columbia Gas System Service Corporation, 1600 Dublin Road, Columbus, OH 43216)

A method is proposed for measurement of the elastic and damping properties of soils in the 100- to 1000-Hz frequency range. The soils are assumed to behave as linear viscoelastic materials. The measurement technique involves exciting vibration of a soil sample through fixtures of known dynamic characteristics. The mechanical impedance matrix of a system composed of sample and fixtures is measured directly. A model of this combined system permits the mechanical impedance matrix of the soil sample alone to be extracted. From this the dynamic properties of the soil material may be computed.

1:15

UU2. Comparison of complex dielectric constant with dynamic elastic moduli of several elastomers. J. Burns (Florida Institute of Technology, Melbourne, FL 32901) and P. S. Dubbelday (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337)

Complex dielectric constants were measured for several types of rubbery elastomers over a temperature range from -10 to 30°C . The samples included a series of neoprenes with the same composition except for variable amounts of carbon black filler from 0% to 33%. These dielectric data are compared with elastic moduli measured on similar material to determine which elastic response is most closely correlated with dielectric properties. The correlation is discussed in terms of physical models of the polymers. [Work supported by ONR.]

1:30

UU3. Development of a "dry" system for the measurement of porosity. Yvan Champoux (Mechanical Engineering Department, Faculty of Applied Science, Université de Sherbrooke, Sherbrooke, Quebec J1K 2R1, Canada), Michael R. Stinson, and Gilles A. Daigle (Physics Division, National Research Council, Ottawa, Ontario K1A 0R6, Canada)

In this paper, an experimental system that permits rapid and convenient measurement of the porosity of materials is described. The use of liquid, either directly in the technique or for temperature stabilization, is avoided. In this sense, it is truly a dry porosity measurement technique. Porosity is one of several physical parameters required for various theoretical models, to characterize a porous material. The proposed technique is based on the approach suggested by Beranek [J. Acoust. Soc. Am. **13**, 248 (1942)]. The pressure change in a closed volume (with and without a sample) is monitored for a given change in volume. To simplify the ma-

nipulation and to improve the precision, the liquid is replaced by a calibrated plunger and an electronic pressure transducer. The volume displacement being kept small, the temperature rise is minimal and the isothermal condition is obtained without the need to use any liquid-filled bath to stabilize the temperature. A calibration procedure is required though, to account for possible movement of the transducer diaphragm. Preliminary experimental results and error analysis suggest that porosity can be measured with this approach with 1%-2% accuracy.

1:45

UU4. Surface phenomena of finite amplitude, pure-tone sound on air-saturated porous materials. Jason D. McIntosh (Department of Electrical Engineering, University of Minnesota, Minneapolis, MN 55455)

It is well known that the flow resistance, as defined by Darcy's law, becomes a function of the fluid flow for large fluid velocities. For porous materials in air, there appears to be two separate nonlinear flow regimes; for low velocities, the resistance coefficient increases with the square of the fluid velocity; and for high velocities, the resistance coefficient increases linearly with the magnitude of the fluid velocity. In this paper, the two nonlinear flow regimes are parametrized for several porous materials. Nonlinear corrections are made to linear wave theory and numerical solutions for pure tones are obtained that then are used to predict surface admittance, absorption, and internal pressure attenuation for infinite and finite length samples. Among the phenomena predicted and also observed experimentally is a peaking and then a rolling off the surface absorption of a finite length sample as the surface pressure is increased. Also, a surface pressure criterion is put forth that gives an approximate maximum surface pressure that a porous material could be exposed to before these nonlinear phenomena become important in describing the material's characteristics. [This work was supported in part by a grant from the Defense Systems Division of Honeywell, Incorporated.]

2:00

UU5. Wave attenuation in 1-3 piezoceramic composites due to electric conductivity. X. Q. Bao, Vijay K. Varadan, and Vasudara V. Varadan (Department of Engineering Science and Mechanics, Research Center for the Engineering of Electronics and Acoustic Materials, The Pennsylvania State University, University Park, PA 16802)

Piezoceramic composites can be tailored to provide propagating waves with high attenuation by introducing appropriate electrical conductivity into the materials. Analysis of 1-3 type piezoceramic composite materials is presented. The optimum value of conductivity for maximum attenuation of propagating waves along the PZT fiber direction is discussed. Numerical results for 1-3 PZT/epoxy composites with various volume fractions are shown.

UU6. Acoustic source strength measurements of cavitation noise in a water tunnel. Gerald C. Lauchle (Applied Research Laboratory, Penn State University, Box 30, State College, PA 16804) and Silvio Bistafa (Guaratinguetá Engineering School, State University of São Paulo, Guaratinguetá, 12500-São Paulo, Brazil)

An experimental investigation of the noise generated by cavitation in turbulent shear flow is reported. The cavitation is produced by the flow through a sharp-edged orifice plate mounted in the test section of a 15.25-cm-diam water tunnel. The unique features of this experiment include the direct measurement of acoustic source strength using a reciprocity technique and the development of semi-empirical scaling laws for the noise produced. The source strength measurement method is useful for water tunnel cavitation noise studies because it is nonintrusive; the noise is measured by a reciprocal transducer located outside of the test section, in air. The scaling laws for orifice plate cavitation noise are developed to include the effects of pressure drop across the plate, the mean velocity through the plate, the cavitation number, and the air content level of the water. These scaling formulas were verified by running a similar test in a 30.5-cm water tunnel and comparing the measured source strength spectra to those predicted from the smaller tunnel results. [Work supported by the Applied Research Laboratory.]

2:30

UU7. Scatter in ultrasonic wave attenuation measurements caused by transducers. Mary A. Paul and Anna L. Pate (Department of Engineering Science and Mechanics, Iowa State University, Ames, IA 50011)

The scatter in attenuation of ultrasonic waves in aluminum was investigated with five broadband, nonfocused, immersion transducers with center frequencies of 10 and 15 MHz. Three different experimental techniques were used: two types of multiple echo techniques and a multiple thickness technique. The objective was to investigate the repeatability of the measurement for the same transducer and same method, the bias towards a particular method, and the consistency of the measurements across transducers. No bias toward a particular method was found and repeatability was good. However, the results were found to be transducer dependent. Efforts have currently begun to image the transducers in order to perform their individual characterization. This characterization will be used instead of modeling transducers as rigid pistons. The effect of transducer characterization on the consistency in attenuation measurement will be discussed. [Work supported by Center for NDE, Ames, IA.]

2:45

UU8. Use of gating technique for calibrating sound intensity probes. G. Krishnappa and V. J. Chiu (Engine Laboratory, National Research Council Canada, Ottawa, Ontario K1A 0R6, Canada)

Sound intensity probe calibration in large laboratory spaces using the technique of "time gating" was investigated. In this study, both the sound source and the probes were mounted 2.45 m above the room floor and the nearest reflecting surface was 4.87 m from the measurement point. The source and the probe were separated by 1.8 m. The length of the sound pulse (chirp) and trigger delay were adjusted on an FFT analyzer to prevent the probes from sensing the reflected signal. The performance of the sound intensity probes was compared with sound intensity derived from sound pressure level measurements by a single calibrated microphone. Both the two-microphone probe and the probe incorporating direct measurement of particle velocity were examined at several angles of incidence both in the horizontal and vertical planes. The results demonstrate that the calibration of sound intensity probes can be carried out with reasonable certainty in the 300- to 6000-Hz frequency range and that the "time gating" technique has the advantage of not requiring any expensive special facilities.

UU9. Acoustic source power evaluation based on finite-difference intensity. Ruey-Chang Wei and Anna L. Pate (Department of Engineering Science and Mechanics, Iowa State University, Ames, IA 50011)

Radiated power of a baffled piston was investigated by numerical simulation and also by measurements in an anechoic chamber with the use of the finite-difference intensity. Several parameters involved in the acoustic power evaluation were investigated. These include: the type of the scanning surface (hemisphere and half-cube), the location of the scanning surface, source directivity ($ka = 1$ and $ka = 10$), microphone spacing, and the number and density of the measurement points. As an independent experimental check, a pressure-based, farfield method was used to evaluate the source power. The source power estimated from the finite-difference intensity was within 1-dB agreement with the results from the farfield pressure based method. As a result of this study the effects of each parameter on the accuracy can be estimated, and consequently the choice of these parameters can be made. [Work supported by NSF.]

3:15

UU10. Shifts in transverse vibration modes of elastic beams resulting from a discontinuity in beam stiffness. Lawrence M. McClure and Robert D. Finch (Department of Mechanical Engineering, University of Houston, Houston, TX 77204-4792)

Vibration characteristics of elastic beams are modified by a change in stiffness due to a discontinuity. This effect was first investigated by W. T. Thompson [J. Appl. Mech., Trans. ASME 16, 203-207 (1949)]. In this paper there is a report on a study of the frequency shifts due to variations in individual parameters of a crack in a beam. A quantity termed the signature sum is defined in terms of deviation from the modal frequencies of an uncracked beam and is proposed as an index to identify the presence of a discontinuity in a beam under investigation. [Work supported by NSF.]

3:30

UU11. An omni-directional, ultrasonic acoustical source for three-dimensional position monitoring in air. Jack Leifer and Ilene J. Busch-Vishniac (Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712)

An acoustically based means of tracking the position of the end effector of a moving industrial robot arm has been proposed. The system consists of an omni-directional acoustical source fixed to the robot arm, and several stationary sensors positioned throughout the robot's workspace. Based upon the time-of-flight of a 500-kHz signal traveling from the source to each sensor, the position of the end effector can be determined. This scheme is potentially an effective, yet economical alternative to the more complex optical tracking systems currently in use. The acoustical source incorporates an exponentially tapered solid Webster's horn that is driven in resonance by a commercial piezoelectric transducer mounted on its wide end. A high-amplitude vibration is produced at the narrow end of the horn, where the resulting sound field can be described by the piston equations. The pressure and omni-directional characteristics of the acoustical field produced are enhanced beyond those that could be generated by a simple flat piezoelectric source alone. Results obtained with this source confirm the feasibility of using ultrasound in air as a means of accurately determining positions in three dimensions. [Work supported by General Motors Research Laboratories.]

UU12. Generation of acoustic pulses of short duration from laser pulses on metal foils. Peng Jiang and Robert E. Apfel (Department of Mechanical Engineering, Yale University, New Haven, CT 06520)

To calibrate transducers in water, a laser was used as a means to generate acoustic pulses of short duration. In the experiments, samples were chosen from the available metal foils (stainless steel and aluminum) with different thicknesses. Several conclusions can be drawn from the experiments. (1) The energy density of the incident laser pulse has a strong effect, depending on the properties of sample, on the duration of the generated acoustic pulses. (2) Thermal expansion β and thermal diffusivity length L are two important parameters that affect the generated acoustic waveforms, where $L = (2Dt_p)^{1/2}$ and $D = K/c_p\rho$, K is the thermal conductivity, c_p is the specific heat, ρ is the density, and t_p is the duration of a laser pulse. (3) If the thickness of a sample is comparable to the thermal diffusivity length L , it will affect the generated waveform. (4) Laser generation of acoustic pulses from metal foils may be able to be used to calibrate receiving transducers in a liquid, at least in a relative sense. [Work supported, in part, by U.S. National Institute of Health through Grant 1R01CA39374.]

4:00

UU13. Acoustical activity in structurally chiral composites. S. K. Yang, V. V. Varadan, and V. K. Varadan (Department of Engineering Science and Mechanics, Research Center for the Engineering of Electronic and Acoustic Materials, The Pennsylvania State University, University Park, PA 16802)

"Optical activity" exhibited through electromagnetic wave propagation and scattering has been well-studied in recent years even at GHz frequencies. Acoustical activity, on the other hand, was first experimentally observed in 1968 by Portugal and Burstein [Phys. Rev. 170, 673-678 (1968)]. Lakhtakia *et al.* [paper submitted for publication] argued that since electromagnetic waves can discriminate the chirality or handedness of the microstructure, there is no reason why other transverse waves cannot do so also. In general, the fields in an elastic solid consist of both transverse and longitudinal components. Therefore, an elastic solid can exhibit a phenomenon similar to optical activity that has been referred to as acoustic activity by some authors. To verify this hypothesis, structural chirality induced by intrinsically anisotropic materials was studied. Four laminated composite samples were prepared. In two of the samples, the fiber direction was rotated from one layer to the next so that the resulting sample was either left or right handed. In the third sample, the fiber directions were random from layer to layer, whereas in the fourth they remained aligned resulting in a sample with orthotropic symmetry. These samples were then tested by using shear wave transducers in an ultrasonic through transmission experiment. Our results indicate that shear wave transducers are able to sense the handedness of the samples and detect additional shear modes.

4:15

UU14. Forward projection of a transient tomographically reconstructed pressure field. II. Michael J. Forbes,^{a)} Stephen V. Letcher (Department of Physics, University of Rhode Island, Kingston, RI 02881), and Peter R. Stepanishen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881)

The forward projection of transient pressure fields from ultrasonic transducers has been investigated both experimentally and theoretically. The transient pressure field of a plane in front of a transducer is projected to another plane further from the transducer using wave-vector and time-domain methods that are implemented using FFT's. We have previously shown that this projection algorithm gives good results when applied to the field of a piston that is calculated using impulse-response methods. In the current work, experimentally determined transient pressure fields in a plane in front of typical medical or NDE transducers have been obtained using optical tomographic methods to measure a two-dimensional slice of the field as a function of time. The forward projection of the space- and time-dependent field via our projection algorithm yields results that are in good agreement with the optically measured field at the new plane. A comparison of analytical and experimental results will be discussed along with potential applications of the method for transducer calibration and characterization. ^{a)} Present address: Naval Underwater Systems Center, Newport, RI.

4:30

UU15. Unidirectional magnetostrictive/piezoelectric hybrid transducer. John L. Butler (Image Acoustics, Inc., P. O. Box 6, North Marshfield, MA 02059), Stephen C. Butler (Massa Products Corporation, 280 Lincoln Street, Hingham, MA 02043), and Arthur E. Clark (Naval Surface Weapons Center, Silver Spring, MD 20910)

The object of the hybrid transducer is to produce enhanced motion at one end and canceled motion at the other end. In the simplest form the transducer consists of a quarter wavelength section of piezoelectric material joined to a quarter wavelength section of Terfenol-D rare-earth magnetostrictive material. A 3-kHz experimental hybrid transducer with a total length of 18 in. was constructed with a small central mass and two small equal end piston masses. Measurements on this device showed an average front-to-back ratio of approximately 17 dB. The operating theory, equivalent circuit analysis, and measured results will be presented. The results confirm the original 1984 theoretical prediction [U.S. Patent 4,443,731]. [Modeling effort supported by SBIR.]

4:45

UU16. Improvement of reflection and transmission coefficient measurements on test panels. C. Audoly and C. Giangreco (Groupe d'Etudes et de Recherches de Détection Sous-Marine, Le Brus, 83140 Six-Fours, France)

The efficiency of acoustic materials is often evaluated by measuring the reflection and transmission coefficients on test panels in tanks. The aim is to obtain the intrinsic characteristics of the material, as if the panel were infinite. However, when the size of the panel is not large compared to the wavelength in water, the measurements are disturbed by scattering phenomena. Different techniques (use of surface hydrophones and transmitters, intensity receivers, double-layer transducers) are studied experimentally and theoretically and their use to improve the measurements is examined. For normal incidence measurements, good results are obtained with a surface hydrophone placed in the nearfield of the test panel.

Session VV. Structural Acoustics and Vibration VI: Wave Propagation in Structures

Martin J. Pechersky, Chairman

Department of Engineering Science and Mechanics, Applied Research Laboratory, Pennsylvania State University, University Park, Pennsylvania 16802

Chairman's Introduction—1:00

Contributed Papers

1:05

VV1. Energy flow associated with fundamental elastic wave types on inhomogeneous cylindrical shells. Allen D. Pierce (Graduate Program in Acoustics, Pennsylvania State University, 157 Hammond Building, State College, PA 16804)

The basic shell dynamic equations for an inhomogeneous cylindrical shell (slowly varying radius, thickness, and material properties) result from Hamilton's principle. The version adopted here is one which yields the Donnell equations in the limiting case of a homogeneous thin shell, although the same technique applies for more sophisticated shell theories. There are three wave types that result from the theory that can be identified in the high-frequency limit as compressional (longitudinal), in-plane shear, and flexural. The propagation of these waves is directionally dependent, frequency dependent, and dispersive. The general ray acoustics formulation developed in previous papers by the author, by Felsen and Lu, and by Norris is extended to include the proper invariants for propagation along ray tubes. The energy density and energy flux vectors for each of the three wave types are derived with the aid of Hamilton's principle and a Poynting's theorem expressing conservation of wave energy is obtained for each wave type. This identification allows the discussion of the energy transfer when waves are partially reflected and converted to other wave types at ribs or at abrupt transitions of shell radius. [Work supported by ONR and by the William E. Leonhard endowment to Pennsylvania State University.]

1:20

VV2. Surface waves in layered structures with application to vocal-fold vibration. Richard S. McGowan (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

The initiation of surface waves over a layered structure, including an air channel, a thin elastic layer, an inviscid waterlike fluid, and an impedance boundary, is discussed. This structure is a rudimentary picture of the vocal folds near the edges. The initiation of surface waves at the interface between the air and elastic layer is similar to the problems of initiation of wind waves over water and the flutter of panels. Early linear analyses of surface wave initiation considered Kelvin-Helmholtz instability, a static instability, and the use of Jeffreys' sheltering coefficient as a destabilizing mechanism for a dynamic instability. Both static and dynamic instabilities are possible in vocal-fold vibration. However, the dynamic type of instability is the mechanism that usually is operable in a layered structure model of vocal-fold vibration, although Jeffreys' sheltering coefficient does not provide the correct term for negative damping. This is confirmed by the lumped element two-mass model, a very successful model for simulating some aspects of vocal-fold vibration [K. Ishizaka and J. L. Flanagan, *Bell Syst. Tech. J.* **51**, 1233 (1968)]. The two-mass model also illustrates how to incorporate dynamic pressure loss into the layered structure approach to account for the negative stiffness. The layered structure approach has the advantage of allowing for testing of the effects of changing the mechanical properties of various histological layers on the initiation of the surface wave. [Work supported by NIH grants HD-1994, NS-13870, and NS-13617 to Haskins Laboratories.]

1:35

VV3. Analysis of kinetic energy repartition for in-plane beam structures. J. B. Piau, T. Loyau, and J. Nicolas (Groupe d'Acoustique de l'Université de Sherbrooke, Département de Génie Mécanique, Université de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada)

In this paper, kinetic energy repartition for in-plane beam structures is analyzed using three different methods: the statistical energy analysis, the energy influence coefficient method (based on the modal approach), and the exact method (based on the wave approach). The two last methods allow the definition of a linear relation between the average kinetic energy of each beam and the excitations applied to all the structure. Therefore the contribution of each excitation to the kinetic energy of each beam can be shown. In the structure, the beam angle at a junction can take any value. A junction model with three degrees of freedom is developed to account for longitudinal and flexural motions. A comparison between the three methods is performed. The analysis results are presented for different angles, different types of junctions, and supports.

1:50

VV4. Critical frequencies of ribbed plates. G. Maidanik, J. Dickey (David Taylor Research Center, Annapolis, MD 21402-5067), and J. Ertel (Physics Department, United States Naval Academy, Annapolis, MD 21402-5026)

The vibrational and radiative properties of panels are often cast in terms of the relationship between the two sets of properties. The relationship is expressed in terms of the radiation efficiencies of these panels. Often interest lies in the dependence of the radiation efficiency on frequency; the radiation efficiency is, therefore, usually displayed as a function of frequency. If the panel is basically a plate responding in flexure it is customary to ascertain the radiation efficiency as a function of the normalized frequency (ω/ω_c), where ω is the frequency variable and ω_c is defined as the frequency at which the phase velocity of the free waves, in a specified direction, on the unloaded uniform plate matches the speed of sound in the irradiated fluid. The critical frequency may have directional properties. However, if the uniform plate is isotropic, the critical frequency is independent of direction. When the plate is nonuniform; e.g., because the plate is ribbed and/or finite, the "free waves and their phase velocity" on the unloaded nonuniform plate may not correspond to those on the unloaded uniform plate. Even if these free waves admit to the definition of the critical frequency ω_c , this critical frequency may not be equal to ω_c . If there is merit in displaying the radiation efficiency as a function of the normalized frequency, which critical frequency need be used in the normalization? The paper is a brief attempt to answer this question. The definition of the critical frequencies and their use in normalizing the frequency for displaying the radiation efficiencies of regularly ribbed plates are cited as examples.

VV5. Propagation of elastic waves on thin-walled circular cylinders. Hyun-Gwon Kil (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332), Allan D. Pierce (Department of Mechanical Engineering, Pennsylvania State University, University Park, PA 16802), and Jerry H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

This paper examines the nature of elastic waves propagating in a thin-walled circular cylinder from the viewpoint that the cylinder's geometry is a source of anisotropy. A point-excited circular cylinder is regarded as a two-dimensional anisotropic medium extending over an infinite domain formed from the stretched angular coordinate. Correspondingly, the displacement field is represented as a superposition of waves propagating out from localized sources into an unbounded medium. The excitation of waves by various types of point force is analyzed using Lighthill's theory. This leads to an expression for the farfield response of the point-excited unbounded medium, which can be expressed in a simple asymptotic form at frequencies somewhat higher than the ring frequency. Thus, in this frequency range, the farfield response of a point-excited circular cylinder can be easily obtained using the corresponding asymptotic farfield response of an unbounded medium. Using these results, it is shown that the wave motion in a thin-walled circular cylinder at moderate frequencies may be interpreted in terms of the wave motion in a thin plate. [Work supported by ONR.]

2:20

VV6. Theory and measurements of transient ultrasonic waves in a viscoelastic plate. Richard L. Weaver (Department of Theoretical and Applied Mechanics, University of Illinois—Champaign—Urbana, Urbana, IL 61801), Wolfgang Sachse, and Lin Niu (Department of Theoretical and Applied Mechanics, Cornell University, Ithaca, NY 14853)

The nearfield responses of a thick viscoelastic plate are computed for epicenter, off-epicenter receiver locations to a point step load acting normal to the surface. Analysis of these responses shows that the ray arrival behavior of a transient ultrasonic signal propagated through a thick viscoelastic plate can be analyzed to recover the plate material's dispersion and attenuation. Such measurements form the basis of the *point-source/point-receiver* measurement technique. Signal processing algorithms are demonstrated with synthetic and real signals obtained via this technique in elastic and viscoelastic materials.

2:35

VV7. Forced vibrations of stiffened cylindrical elastic panels. Zahit Mecitoglu and M. C. Dökmeci (Faculty of Aeronautics and Astronautics, Istanbul Technical University, Maslak, Istanbul 80626, Turkey)

The objective of this paper is to investigate the forced response of a stiffened circular cylindrical panel within the frame of Love's first approximation theory of elastic shells. The effects of stiffeners are taken into account by the orthotropic material approach, including the eccentricity of stiffeners. The variational equation of motion for the forced vibrations of thin elastic shells with clamped edges is derived from Hamilton's principle. The stiffened elastic panel is discretized by the Semiloof elements. Then, with the help of the variational equation, the matrix equation is obtained to determine the frequencies and mode shapes of the stiffened circular cylindrical elastic panel. Two types of stiffened shell panels are considered for forced-response analysis: the stiffened models by the stiffeners with rectangular cross section and with profile shape cross section. The influence of the spacings and dimensions of stiffeners on the forced response of the stiffened circular cylindrical panel is examined. The numerical results are plotted with respect to the parameters, and they are compared with certain earlier results. Further needs of research are pointed out.

THURSDAY AFTERNOON, 25 MAY 1989

REGENCY A, 1:25 TO 5:00 P.M.

Session WW. Physical Acoustics VIII and Animal Bioacoustics I: Application of Physical Acoustics to Agriculture

Lawrence A. Crum, Chairman

National Center for Physical Acoustics, University of Mississippi, University, Mississippi 38677

Chairman's Introduction—1:25

Invited Papers

1:30

WW1. Acoustical detection of insect larvae in post-harvest commodities. J. C. Webb (USDA, ARS, Insect Attractants, Behavior, and Basic Biology Research Laboratory, Gainesville, FL 32604)

An acoustical system to detect insect larvae in fruit, nuts, and grain will be described. The system can detect 1-day-old fruit fly larva in grapefruit, mangos, or other similar fruit and lepidopterous and coleopteran larva in individual kernels of grain as well as bulk grain. Two types of detectors have been developed for this work. One type is based on the closed organ pipe principle that works well with both fruit and grain and a piezoelectric sensor that works primarily with grain. The signal of the feeding larvae is detected, amplified approximately

100 dB, filtered through the appropriate bandpass filters, and then stored on computer. The computer and software are capable of monitoring the larva 100% of the time through its life cycle. The system has applications both in basic research and quarantine inspection of insects in import and export commodities.

2:00

WW2. Acoustic communication in pest mole crickets. T. G. Forrest (National Center for Physical Acoustics, P. O. Box 847 Fraternity Row, University of Mississippi, University, MS 38677)

Mole crickets are common pests of turfgrass and pastures in the southeastern United States. The males produce species-specific calling songs that function to attract mates. Males begin calling after sunset from a burrow constructed prior to calling. The burrow's opening expands exponentially and the burrows are "tuned" during their construction. Output increases 20 dB during the tuning process. Sound fields are hemispherical. Power output ranges from 4 to 20 μ W and is dependent upon the size of the male and the moisture of the soil surrounding the burrow. Flying females respond to the calling song and land near calling males. Males calling in an outdoor arena were highly variable in the number of individuals attracted to their calling songs. Intensity differences among the males was the major determinant in the response of females to males in the arena. The differential response to calling males can be explained using a simple model of pressure fields surrounding two sources.

2:30

WW3. Sound and the moths that infest bee hives. Hayward G. Spangler (USDA, ARS, Carl Hayden Bee Research Center, 2000 E. Allen Road, Tucson, AZ 85719)

Two moths that infest bee hives, the lesser wax moth, *Achroia grisella*, and the greater wax moth, *Galleria mellonella*, generate ultrasonic pulses by buckling tymbals. Lesser wax moth males call continuously, producing 100-kHz sound pulses by which females orient and find them. Greater wax moth males generate a short series of 75-kHz sound pulses, but females find them by orienting to male pheromone. Females wing fan in response to male sound and may signal males to release more pheromone to guide them. In contrast, each tymbal of the related rice moth, *Corcyra cephalonica*, has nine striae that buckle and recover in sequence to produce 125-kHz sound in an irregular train of pulses. Female moths may not distinguish 7.5-ms pulse trains from 7.5-ms bursts of continuous sound. Both lesser wax moths and rice moths can be located and identified by detecting the sound of calling males. Acoustical traps may prove useful for monitoring or controlling populations.

3:00

WW4. Acoustical peak identification system (APIS): An acoustical technique for distinguishing between European and Africanized honeybees. Howard T. Kerr and M. E. Buchanan (B-Tec, Inc., Route 11, Box 7, Maryville, TN 37801)

A simple method has been developed to distinguish between European and Africanized honeybees on the basis of differences in the spectral content of sounds made by worker bees in flight. The spectral content of the flight sounds for each subspecies may generally be correlated to characteristic taxonomical features (i.e., wing size, body weight, thorax dimensions, etc.) and physical behavior (i.e., foraging, stinging, etc.). Acoustic signals from over 800 individual flying European and Africanized honeybees were collected. A time-averaged signal was generated for each specimen, and standard Fourier analysis methods were used to generate power spectral density plots. Examination of the PSD's revealed that distinguishing differences exist between the acoustical signatures of flying European honeybees and flying Africanized honeybees. Specifically, the flying European honeybee signature has a fundamental power peak in the 210- to 230-Hz frequency range whereas the flying Africanized honeybee signature has a fundamental power peak in the 260- to 280-Hz frequency range. This substantial difference is easily embodied in a variety of simple identification algorithms. A patent application was submitted for the acoustical peak identification system (APIS), and a simple, inexpensive, field-portable instrument based on acoustical signature techniques has been developed for screening identification of the Africanized subspecies of honeybees. This instrument is called "Buzz Buster" and is being marketed by B-Tec.

WW5. Audio spectra and related behavior of *Monochamus titillator* F. larvae. R. T. Walden (National Center for Physical Acoustics, University, MS 38677) and T. E. Nebeker (Department of Entomology, Mississippi State University, Mississippi State, MS 39762)

The behavior and audio spectra of *Monochamus titillator* F. larvae, the southern pine sawyer, were observed and measured in its natural environment. The sawyer is being utilized to assist in locating southern pine beetle infestations. The sawyer's audio emission is primarily in the 1000- to 2800-Hz range with intensity peaks at approximately 1300 and 2300 Hz. The analysis was obtained using both a sweep-frequency analyzer and an FFT spectrum analyzer. The oscilloscope display gave surprising information in that a single sound output seems to first intensify with time and then rapidly decay. A closer analysis reveals that a single sound is composed of several time decaying but ever intensifying components. The sound repetition rate varies from 30 to 120 pulses per minute depending on environmental conditions. The behavior of the sawyer during its sound production is being studied with regard to season of the year, time of day, ambient temperature, and sound production mechanism.

4:00

WW6. Ultrasonic acoustic emissions from drought-stressed trees: Do bark beetles hear them? Robert A. Haack (USDA Forest Service, 1407 South Harrison Road, East Lansing, MI 48823)

Bark beetles preferentially infest drought-stressed trees for reasons not completely understood. One possibility is that bark beetles are attracted to the ultrasonic acoustic emissions (AEs) that result from breakage of water columns in drought-stressed trees. AEs were measured from 20 to 300 kHz on cut trunk samples and on small, potted trees of several coniferous and hardwood species, using ceramic piezoelectric transducers and a four-channel AE analyzer. Mean rate of emission as well as mean signal duration, rise time, amplitude, and energy were determined at weekly intervals during periods of drying. Some of these results have recently been published [R. A. Haack *et al.*, Florida Entomol. 71, 427-440 (1988)]. Using the mean emission rates and signal parameters to adjust the settings of an ultrasonic pulsing instrument, natural and artificial substrates were pulsed with ultrasound to simulate varying degrees of drought in the presence of bark beetles. The influence of pulsed AEs on the colonization behavior of bark beetles will be discussed. [Work supported by USDA.]

4:30-5:00

Bull Session

THURSDAY AFTERNOON, 25 MAY 1989

ROOM 202, 1:30 P.M.

Meeting of Accredited Standards Committee S3 on Bioacoustics

to be held jointly with the

Technical Advisory Group (TAG) Meeting for ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock

L. A. Wilber, Chairman S3
422 Skokie Boulevard, Wilmette, Illinois 60091

Standards Committee S3 on Bioacoustics. The current status of standards under preparation will be discussed. In addition to those topics of interest, including hearing conservation, noise, dosimeters, hearing aids, etc., consideration will be given to new standards which might be needed over the next few years. Open discussion of committee reports is encouraged.

The international activities in ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock, will be discussed. The Chairs of the TAGs for ISO/TC 43 (H. E. von Gierke), IEC/TC 29 (V. Nedzelitsky), and ISO/TC 108/SC4 (H. E. von Gierke), will report on current activities of these Technical Committees and Subcommittees.

Session XX. Psychological Acoustics VII: Comodulation and Modulation

William M. Hartmann, Chairman

*Department of Physics and Astronomy, Michigan State University, East Lansing, Michigan 48824**Contributed Papers*

1:30

XX1. Phase sensitivity to amplitude-modulated stimuli as a function of component spacing. Savithri Sivaramakrishnan, Glenis R. Long (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907), and Arnold Tubis (Department of Physics, Purdue University, West Lafayette, IN 47907)

The relation between phase sensitivity of a three-component amplitude-modulated (modulation index = 1) stimulus, presented at 40 dB SL, as a function of the frequency spacing of the individual components was examined. Five sets of stimuli with carrier frequency of 2000 Hz and modulation frequencies of 50, 100, 200, 300, and 400 Hz were generated. The phase of the center component was varied from 0–90 deg in steps of 10 deg to give a total of 50 stimuli. Subjects were asked to discriminate the zero phase stimulus from the other stimuli containing the same components in a 3I-2AFC paradigm. The relative pitch, and pitch strength, of stimuli containing the same components at different relative phase was determined using a paired-comparison paradigm. Measurements of each subject's auditory filter characteristics at 2000 Hz (based on the notched-noise paradigm) will be used to interpret the data. [Work supported by a David Ross Fellowship at Purdue University.]

1:45

XX2. Detection and recognition of amplitude modulation with tonal carriers. Stanley Sheft and William A. Yost (Parham Hearing Institute, Loyola University, 6525 North Sheridan Road, Chicago, IL 60626)

The ability of listeners to process multiple sources of sinusoidal amplitude modulation (AM) was evaluated using both detection and recognition procedures. For all conditions, the stimulus was a two-tone (909 and 4186 Hz) complex. Test-AM frequencies were 4 and 17 Hz. In the detection paradigm, d-primes obtained with simultaneous modulation of both tones were compared to the d-primes obtained with modulation of just one of the tonal components. When both tones were modulated at the same frequency, a phase disparity between the envelopes reduced AM detectability. With the envelopes in phase, results showed a linear summation of the d-primes for the individual components. With modulation of the two tones at different AM frequencies, performance approximated optimal processing of two sources of uncorrelated information. A similar result was obtained in the recognition paradigm when the task was to discriminate between the two modulation frequencies. When the AM frequency was fixed and the task was to identify which carrier was modulated, performance was near chance. Results are consistent with processing of near-threshold AM through modulation-specific channels that are broadly tuned to carrier frequency. [Work supported by NINCDS.]

2:00

XX3. Discrimination and identification of modulation rate using a noise carrier. Thomas E. Hanna (Naval Submarine Medical Research Laboratory, Box 900, Naval Submarine Base—New London, Groton, CT 06349-5900)

Modulation-rate perception was measured with three tasks: a fixed-standard, forced-choice discrimination task with a 500-ms interstimulus

interval; a random-standard, forced-choice discrimination task with an 8-s interstimulus interval; and an identification task. Thresholds were obtained for modulation rates from 14–224 Hz using noise carriers bandpass filtered from 500–4000 Hz, 500–1600 Hz, 1700–2800 Hz, and 2900–4000 Hz. The four bands yielded similar results except for modulation rates above 100 Hz, where the 500- to 1600-Hz thresholds were higher. Fixed-standard discrimination thresholds were about 3 Hz for modulation rates up to 66 Hz. Increased thresholds for modulation rates above 66 Hz are presumably due to temporal resolution limits. For modulation rates above 100 Hz, modulation rate resolution with the 500- to 1600-Hz carrier is probably limited by critical band filtering rather than temporal resolution. Resolution in the random-standard discrimination task was similar to that for the identification task. Thresholds were elevated relative to fixed-standard thresholds except at the extremes of the stimulus range. In the random-standard discrimination task, a pronounced criterion bias was also present for stimuli near the extremes of the range. Durlach and Braida's [J. Acoust. Soc. Am. 46, 372–383 (1969)] model describes the data well and provides quantitative measures for modulation-rate perception in good agreement with those for intensity perception. [Work supported by ONR.]

2:15

XX4. Detecting FM of coherently and incoherently modulated components of complex sounds. Robert P. Carlyon (Laboratory of Experimental Psychology, University of Sussex, Brighton BN1 9QG, England)

Thresholds were measured for the detection of FM of four-component carriers, where each carrier component was modulated by the same percentage of its starting frequency ("F0 modulation"). The frequency spacing between components was such that they were resolvable by the peripheral auditory system. Each component was modulated by a single cycle of a 12.5-Hz sinusoid, whose phase was either the same for all components ("coherent modulation"), or differed between components ("incoherent modulation"). Preliminary experiments revealed little or no effect of modulator coherence on thresholds, for stimuli presented either in quiet or in bursts of pink noise. This noise will be compared to the outcome of further experiments, in which the coherent and incoherent signals were presented with competing frequency-modulated sounds. The results of all the reported experiments will be discussed in relation to the mechanisms by which listeners separate out concurrent complex sounds.

2:30

XX5. The effect of envelope energy on the discrimination of cross-spectral envelope phase disparity. Brent W. Edwards and Gregory H. Wakefield (Department of Electrical Engineering and Computer Science, University of Michigan, Ann Arbor, MI 48109)

The envelope cross-correlation difference $\Delta\rho$ can be used to predict thresholds for the discrimination of cross-spectral envelope phase disparity for modulation frequencies up to 64 Hz [Wakefield and Edwards, J.

Acoust. Soc. Am. Suppl. 1 82, S41 (1987)]. For the case of two SAM carriers, $\Delta\rho$ is inversely proportional to depth of modulation which suggests that sensitivity to cross-spectral temporal differences depends on the degree to which carriers are modulated. Results are presented that support the hypothesis that sensitivity is inversely proportional to envelope energy. Thresholds are obtained and $\Delta\rho$'s calculated for SAM carriers of

varied modulation depths. It is found that $\Delta\rho$ for a complex sinusoidal modulator is equal to that for a single sinusoidal modulator when the energies of the two modulators are equal. This relationship is also observed for envelopes of narrow-band noises. In contrast, other measures of envelope fluctuation do not yield constant performance as measured by $\Delta\rho$. [This research was supported by a grant from the AFOSR.]

2:45-3:00
Break

3:00

XX6. Cross-spectral envelope phase discrimination for FM signals. Gregory H. Wakefield and Brent W. Edwards (Department of Electrical Engineering and Computer Science, University of Michigan, Ann Arbor, MI 48109)

Previous measures of cross-spectral envelope phase sensitivity have used amplitude-modulated sinusoidal carriers. To the extent that these measures reflect the fidelity of the auditory signal representation of complex signals, it should be possible to predict phase disparity thresholds for arbitrary classes of signals. Modulation phase discrimination functions were obtained for sinusoidal FM carriers at 500 and 2000 Hz. For a 0-deg phase disparity between the modulators of the standard, sensitivity to a disparity in the phase of FM modulation is monotonically related to the product of modulation frequency and modulation index over the range of modulation frequencies studied (8-64 Hz). A simple model of this discrimination task assumes that FM signals are converted into AM signals through bandpass filtering at the auditory periphery. This model was tested by demodulating the FM signals through auditory filters tuned to 500 and 2000 Hz. Thresholds obtained for these "FM-equivalent" AM signals were nearly an order of magnitude smaller than the original thresholds and it was necessary to "de-tune" the auditory filters (5- to 10-dB slopes) to fit the FM data. [This research was supported by a grant from the AFOSR.]

3:15

XX7. Effects of flanking band proximity, number, and modulation pattern on comodulation masking release. J. W. Hall, III and John H. Grose (Division of Otolaryngology/Head and Neck Surgery, University of North Carolina at Chapel Hill, Chapel Hill, NC 27599-7070)

Comodulation masking release for a 700-Hz pure-tone signal was investigated as a function of the number and spectral positions of 20-Hz-wide comodulated flanking bands. The number of flanking bands ranged from one to eight. The results indicated (1) an effect of proximity, where bands closer to the signal resulted in larger masking release, and (2) an effect of number of bands, where more bands gave rise to larger CMR. Two further sets of conditions were examined, and compared to the condition where all eight comodulated flanking bands were present: In one set of conditions, two of the eight flanking bands were removed; in the other set of conditions, two of the eight flanking bands were replaced with bands that were not comodulated with respect to the other bands. There was very little effect of removing two of the bands, even when close to the signal frequency; however, CMR was substantially reduced when the noncomodulated replacement bands were introduced. These results indicated that the auditory system is not able to disregard modulations on flanking bands that differ from the modulation pattern on the signal band, particularly if these bands are close in frequency. [Research supported by AFOSR.]

3:30

XX8. Signal threshold in frequency-modulated (FM) noise bands as a function of the coherence of modulation. J. H. Grose and J. W. Hall, III (Division of Otolaryngology/Head and Neck Surgery, University of North Carolina at Chapel Hill, Chapel Hill, NC 27599-7070)

In a recent paper reporting the absence of a co-(frequency)-modulation masking release for FM maskers, it was noted that thresholds were lower in maskers comprised of comodulated FM tones than of noncomodulated FM tones [G. S. Schooneveldt and B. C. J. Moore, J. Acoust. Soc. Am. 83, 2290-2292 (1988)]. This was interpreted as being due to a within-channel beating cue. The present study also observed a noncomodulated-comodulated difference but in conditions designed to minimize such a cue. In a baseline condition, detection of a signal was measured in the presence of an on-signal FM noise band. Two flanking FM bands were then added in four configurations relative to the on-signal band: (1) comodulated in both FM and amplitude modulation (AM); (2) comodulated in FM but noncomodulated in AM; (3) noncomodulated in FM but comodulated in AM; and (4) noncomodulated in both FM and AM. The paradigm was tested with sinusoidal and random modulation rates, different depths of modulation and both pure-tone and FM noise band signals. Thresholds for most conditions were raised relative to the respective baseline. However, for both the comodulated and noncomodulated AM conditions, thresholds were usually reduced by the transition from noncomodulated to comodulated FM. [Research supported by AFOSR.]

3:45

XX9. Thresholds for the frequency discrimination of pulsed tones (DLFs) and for the detection of frequency modulation (FMDLs): Effects of noise and random variations in level. Brian C. J. Moore and Brian R. Glasberg (Department of Experimental Psychology, University of Cambridge, Cambridge CB2 3EB, England)

In experiment I, DLFs were measured in four conditions: (1) in quiet with the level fixed at 70 dB SPL; (2) in quiet with the level of each tone varied randomly over a 6-dB range around 70 dB SPL; (3) as 1, but with a continuous bandpass noise designed to mask the upper side of the excitation pattern; and (4) with bandpass noise and with the level of the signal randomized over a 6-dB range. Center frequencies from 0.5-6.5 kHz were used. Except at 6.5 kHz, the results were not consistent with the assumption that the DLFs were based on the detection of changes in excitation level on the low-frequency side of the excitation pattern. Experiment II was like experiment I, except that FMDLs (4-Hz rate) were measured, and, in conditions 2 and 4, a 4-Hz amplitude modulation (AM) with a peak-to-valley ratio of 6 dB was imposed on all signals; the phase of the AM relative to the FM was random. The results were generally not inconsistent with the assumption that the FMDLs were based on the detection of changes in excitation level on the low-frequency side of the excitation pattern. However, neither a single-band nor a multiple-band excitation-pattern model was able to account for the effect of the bandpass noise of the FMDLs. [Work supported by MRC, U.K.]

XX10. Temporal modulation transfer functions for pure tones. William A. Yost and Stanley Sheft (Parham Hearing Institute, Loyola University, 6525 North Sheridan Road, Chicago, IL 60626)

Temporal modulation transfer functions (TMTFs) were obtained for sinusoidally amplitude-modulated (SAM) pure tones. TMTFs were obtained by determining the depth of SAM required for modulation detection in a two-alternative, forced-choice adaptive procedure. TMTFs were obtained for 500-, 1000-, and 4000-Hz carrier frequencies; for durations of 125 and 500 ms (all stimuli were shaped with a 20-ms raised cosine); in gated and continuous background conditions; and for modulation rates ranging from 2–128 Hz. In the gated condition, the carrier tone was gated

on and off and the modulation occurred over the full duration of the tone, while in the continuous condition, the carrier tone was on continuously, and it was modulated during the observation interval. Thresholds for modulation detection were lower in the continuous than in the gated condition and the thresholds were lower for the 500-ms than for the 125-ms stimuli. The TMTFs displayed a bandpass characteristic in all conditions, but the high-pass segment was steeper for the gated than for the continuous conditions. The loss in sensitivity at low modulation frequencies meant that the lowest thresholds were obtained for modulation frequencies of 4–8 Hz in the continuous condition and above 16 Hz in the gated condition. The data will be discussed in terms of other measures of modulation processing and theories of temporal processing, especially those involving envelope extraction. [Work supported by the NINCDS.]

THURSDAY AFTERNOON, 25 MAY 1989

NEWHOUSE II, ROOM 254, 1:30 TO 4:46 P.M.

Session YY. Speech Communication VIII: Word-Level Perception and Production

James R. Sawusch, Chairman

Department of Psychology, SUNY at Buffalo, Amherst, New York 14260

Contributed Papers

1:30

YY1. Syllable structure and units of analysis in speech perception. Jenny DeGroot and Howard C. Nusbaum (Department of Behavioral Sciences, University of Chicago, 5848 South University Avenue, Chicago, IL 60637)

Syllables or their constituents, such as onsets and rhymes, may be basic units in speech perception. If so, then recognition of a target phoneme might be influenced more by its syllabic context (i.e., other phonemes in the same syllable or constituent) than by other aspects of phonetic context. To investigate the influence of syllable structure on phoneme perception, subjects in three response-time experiments were instructed to identify a target phoneme, ignoring variation in an adjacent phoneme. In one condition, the adjacent phoneme was held constant; in a second condition, contextual variation was irrelevant to target identity. In two experiments, context varied either within the same syllabic onset or in an adjacent syllable. The third experiment involved varying an adjacent phoneme either in the same syllable as the target or in a neighboring syllable. If the syllable or onset is perceived as a unit, then varying context within that unit should slow target recognition more than varying context outside the syllable. The results have implications for the role of syllable structure in speech perception. [Research supported by AFOSR and BRSG.]

1:42

YY2. Contingencies in stress patterns and syllabic nuclei: Lessons for searching lexical databases. Mary R. Smith (82 South Pine Avenue, Albany, NY 12208)

In automatic speech recognition programs, attempts have been made to use partial information about the phonetic structure of signals to search large lexical databases. Little attention has been given to the interaction of stress patterns and syllabic nuclei in restricting the searches. A survey of

the MRC Psycholinguistic Database of British English (RP) with 38-K tokens revealed that six stress patterns (out of 17 attested) account for 96% of the tokens in the subset of two- and three-syllable words. Further, three syllabic nuclei (out of 23 attested) account for 50% of the tokens, independently of stress pattern. The interactions of stress pattern and nuclei show not only increased discriminatory power (in information theoretic terms) over either attribute alone, but also reveal significant gaps between expected and observed cases. The gaps, both accidental or principled, further reduce the relevant search spaces and increase the usefulness of this type of partial information over other candidates. [Work supported by Alvey Grant MMI 069.]

1:54

YY3. Malleable frequency effects in spoken word recognition. Paul A. Luce (Department of Psychology, SUNY at Buffalo, Buffalo, NY 14260)

Recent research has demonstrated that word frequency effects in spoken word recognition are primarily due to processes responsible for deciding among multiple lexical items activated in memory and not to differences in activation levels or thresholds of word units. The present study further supports this claim by demonstrating that frequency effects in the identification of spoken words are malleable and can be modified by varying the composition of the experimental stimuli. A set of high-frequency words was presented for identification mixed with either other high-frequency words or with low-frequency words. A set of low-frequency words was also presented with high- or low-frequency words. The results demonstrated that high-frequency words were identified more accurately when presented with other high-frequency words and that low-frequency words were identified more accurately when presented with other low-frequency words. The implications of these results for current models of spoken word recognition, in particular the neighborhood activation model, will be discussed. [Work supported by NIH Grant NS-12179 to Indiana University.]

YY4. Gaining insight into the representational unit for spoken words with a naming task. Deborah A. Gagnon and James R. Sawusch (Department of Psychology, SUNY at Buffalo, Amherst, NY 14260)

Theoretical proposals for the representational unit used in recognizing spoken words include the phoneme, allophone, and position-specific phoneme. The present experiment was devised to differentiate among these proposals. Subjects heard natural CVC stimuli in prime-target pairs and named the second item (target) as quickly as possible. The phonetic overlap between prime and target was varied to create five different intra-pair relations such that the expected pattern of priming effects across the set of relations was different for each of the three phonetic representations. The obtained pattern is inconsistent with an abstract phonemic unit and points to either a position-specific phonemic or allophonic unit. This pattern was found for both word and nonword stimuli. Thus these priming results appear to tap the recognition process prior to lexical activation. Priming effects were eliminated when stimuli within a prime-target pair were produced by different voices. The implications of the findings for refining models of spoken word recognition will be discussed. [Work supported by NINCDS.]

2:18

YY5. Interword coarticulation modeling for continuous speech recognition. Mei-Yuh Hwang, Hsiao-Wuen Hon, and Kai-Fu Lee (Computer Science Department, Carnegie-Mellon University, Pittsburgh, PA 15213)

In large-vocabulary continuous speech recognition, subword units must be used for practical reasons. Context-dependent phone models have become a very successful class of subword units. These phone-sized models take into account the neighboring phonetic contexts, which strongly affect the realization of a phone. However, previous approaches have only considered intraword coarticulation, and have ignored interword coarticulation, which is very important in continuous speech, especially for short function words like "the" and "a." This study extends triphone-based modeling to interword coarticulation modeling. A simple extension of triphones is problematic due to the sharply growing number of triphones. In order to contain this growth, a maximum-likelihood clustering procedure was introduced to reduce 7057 intraword and interword triphones to 1000 generalized triphones. Interword generalized triphones were incorporated into a large-vocabulary, speaker-independent, continuous speech recognizer, SPHINX [K. F. Lee and H. W. Hon, *Large Vocabulary Speaker-Independent Continuous Speech Recognition* (ICASSP, 1988)]. This improvement reduced the number of errors by as much as 44% on the 1000-word DARPA resource management task. This demonstrates the importance of interword coarticulation modeling, and the effectiveness of the methods used.

3:10

YY8. Consequences of new and old information on the production of grammatical categories. Jan Charles-Luce (Department of Communicative Disorders and Sciences, SUNY at Buffalo, Buffalo, NY 14260)

A recent study [C. A. Fowler and J. Housum, *J. Mem. Lang.* 26, 489-504 (1987)] demonstrated that talkers distinguish second (old) token productions from first (new) token productions through shortening. Moreover, listeners could use this information to identify old from new tokens. Of interest for the present investigation was the finding that topic

YY6. The Delta System, version 2: A tool for testing phonological and phonetic theories. Susan R. Hertz (Eloquent Technology, Inc., P. O. Box 4328, Ithaca, NY 14852-4328 and Phonetics Laboratory, DMLL, Cornell University, Ithaca, NY 14853)

The Delta System is a software system that lets linguists test the relationships of phonological and phonetic units through speech synthesis [J. Acoust. Soc. Am. Suppl. 1 75, S60 (1984)]. The Delta programming language now has an improved syntax with greater appeal to linguists, along with a variety of new features. For example, it has new constructs for manipulating and testing Delta's multilevel utterance representation, for preventing groups of rules from crossing particular linguistic boundaries, and for specifying the timing of phonetic events with respect to higher level linguistic units. The Delta debugger now lets users step through their programs statement by statement, and interactively modify their utterance representation on the fly (for example, to hear the effect of a longer duration). A macroprocessor lets users tailor the syntax of Delta programming language statement and debugging commands to their liking. The system compiles Delta programs into C programs and lets users intermingle Delta and C statements. The system will be available for demonstration.

2:42

YY7. A new approach to English text-to-phoneme conversion using Delta, version 2. Susan McCormick (Eloquent Technology, Inc., P. O. Box 4328, Ithaca, NY 14852-4328) and Susan R. Hertz (Eloquent Technology, Inc., P. O. Box 4328, Ithaca, NY 14852-4328 and Phonetics Laboratory, DMLL, Cornell University, Ithaca, NY 14853)

The Delta System, version 2 is being used to develop a new set of text-to-speech rules for English. This paper focuses on the text-to-phoneme portion of these rules. These text-to-phoneme rules are based on an earlier rule set developed by Hertz [J. Acoust. Soc. Am. Suppl. 1 69, S83 (1981)], but are much more elegant and complete, taking advantage of Delta's flexible facilities for building multilevel utterance representations and matching patterns against them. The rules convert an input text into an utterance representation containing the lexical, morphological, and phonological structure of the utterance. Sophisticated algorithms are used for identifying prefixes (productive and nonproductive ones), suffixes, and roots (including the roots of compound words), for predicting lexical stress, for assigning grammatical categories, and for predicting pronunciations. Portions of these algorithms will be described, with a focus on how the Delta programming language allows them to be expressed succinctly and clearly. The rules will be available for demonstration.

2:54-3:10
Break

words shortened relatively less. Thus talkers appear to treat certain types of information differentially. To investigate this idea further, the present study examined four grammatical categories and a fifth topic noun category when these were produced as old and new information. To determine talkers' sensitivity to new versus old information, two passage types were constructed: coherent (paragraph of related sentences) and noncoherent (paragraph of unrelated sentences). The results show that duration does not decrease from new to old productions for all grammatical categories. However, this finding is not independent of passage type. Implications of these results for talkers' sensitivity to semantic information will be discussed. [Work supported by NIH Grant NS-12179 to Indiana University.]

YY9. On-line measures of comprehension of natural and synthetic speech. James V. Ralston, John W. Mullennix, Beth G. Greene, and Scott E. Lively (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Accumulated perceptual research comparing natural and synthetic speech indicates relatively large differences in tasks assessing acoustic-phonetic processing, and small differences in tasks assessing higher levels of processing related to comprehension. Studies comparing comprehension of passages of fluent natural and synthetic speech have generally examined performance on questions presented after subjects have listened to a passage. Such postperceptual measures are known to be relatively insensitive to differences in "real-time" processing operations. The present investigation employed an "on-line" measure of processing, i.e., word monitoring, to study comprehension. Subjects in these studies were presented with three types of passages—(1) natural speech, (2) high-quality synthetic speech, and (3) low-quality synthetic speech—and were required to monitor for target words as well as verify postperceptual comprehension questions. Monitoring latencies and verification performance will be discussed in terms of differences in perceptual processing and comprehension of natural and synthetic speech. [Work supported by NSF.]

3:34

YY10. The effects of perceptual learning on capacity demands for recognizing synthetic speech. Lisa Lee and Howard C. Nusbaum (Committee on Cognition and Communication, 5848 South University Avenue, The University of Chicago, Chicago, IL 60637)

Previous studies have demonstrated that recognition of synthetic speech improves with moderate amounts of training. Three experiments were performed to determine whether the effects of training are due to changes in the attentional capacity required to process the speech. Attentional capacity available for recognizing speech was manipulated using a recall task with different length lists of visually presented numbers. In two experiments, before and after training, the primary task required subjects either to recall lists of words or to perform speeded word recognition for synthetic speech. Whereas training improved performance with synthetic speech, decreasing available capacity impaired performance. In a third experiment, the intelligibility of the speech was manipulated by using speech from different text-to-speech systems. The results of these experiments suggest that neither training nor intelligibility affects the amount of attentional capacity required to recognize synthetic speech. Rather, as perceptual learning occurs, listeners begin to allocate their attentional resources more efficiently to the processing of synthetic speech. [Work supported by AFOSR and BRSG.]

3:46

YY11. Prosodic information in telephone digit strings. Sara H. Basson, Judith Spitz, and Charles R. Jankowski (NYNEX Science and Technology, Artificial Intelligence Speech Technology Group, 500 Westchester Avenue, White Plains, NY 10604)

Intonation of naturally produced telephone digit strings appears to conform to particular patterns. Concatenating strings of digits produced with "default" intonation highlights the importance of maintaining appropriate intonation patterns for intelligible, natural-sounding speech. The purpose of this experiment was to quantify the acoustic events resulting in natural-sounding telephone number digit strings. Acoustic features of approximately 1000 digits produced in the context of telephone digit strings were measured. The data were gathered at MIT by presenting volunteer speakers with lists of seven-digit numbers to read. Shifts in fundamental frequency of the vowel, changes in overall energy, and digit duration as a function of position in the digit string were calculated. These results can prove useful for speech recognition and generation. For auto-

matic speech recognition of digit strings, prosodic contours provide cues about word boundary locations and serve as a source of information for error detection and correction. When concatenating stored speech to produce natural sounding digit strings, these findings provide guidance as to how many templates need to be created. Finally, for text-to-speech synthesis, these results suggest the number of rules necessary to model natural-sounding digit strings.

3:58

YY12. Comparing a magnitude estimation technique and a pair comparisons technique for use in assessing quality of text-to-speech synthesis systems. Chaslav Pavlovic (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, Iowa 52242 and Centre National d'Etudes des Telecommunications, BP 40, 22301 Lannion, France), Christel Sorin, Jean Pierre Roumiguere, and Jean Pierre Lucas (Centre National d'Etudes des Telecommunications, BP 40, 22301 Lannion, France)

This study was designed to compare a magnitude estimation procedure and pair comparisons procedure for use in scaling quality of text-to-speech synthesis systems. To this goal, the psychophysical scale values that result from the magnitude estimation procedure were compared to the values obtained from a pair comparisons procedure. Four different synthesis systems and three different prosody rules in all possible combinations were scaled using a magnitude estimation technique. To obtain the pair comparisons scale values, the results reported by Lucas *et al.* [CNET report NT/LAA/TCS/375, Tom I, Lannion, France (1988)] for the same systems were used. They were reanalyzed for this purpose. The data indicate a good general agreement between the results of the two experiments. Thus the two methods seem to validate each other. The relative efficiency of the magnitude estimation technique is greater than that of the pair comparisons technique.

4:10

YY13. Perceptual normalization of talker differences. Howard C. Nusbaum and Todd M. Morin (Department of Behavioral Sciences, 5848 University Avenue, The University of Chicago, Chicago, IL 60637)

Different talkers may produce the same phoneme with different acoustic patterns and different phonemes with similar acoustic patterns. In order to recognize speech correctly, a listener must use information about the talker's vocal characteristics. A speeded target-monitoring task was used to investigate whether each token of speech contains sufficient information for normalization or whether a sample of a talker's speech must be heard to normalize subsequent segments. Subjects listened to a sequence of syllables for a particular target. In one type of trial, targets and distractors were produced by a single talker; in other trials, targets and distractors were produced by a mix of talkers. If each token of speech is self-normalizing, there should be no difference between conditions. Otherwise, performance should be worse in mixed-talker trials. In another experiment, the available cognitive capacity for recognizing speech was manipulated and recognition of speech produced by a single talker compared to a mix of talkers was examined. The results of these experiments have implications for the type of mechanisms used in normalization and for the attentional limitations they impose on listeners. [Work supported by AFOSR and BRSG.]

4:22

YY14. Detailing the nature of talker variability effects in speech perception. John W. Mullennix and David B. Pisoni (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Previous studies have shown that changes in a talker's voice from stimulus to stimulus affect the perception and recall of spoken words [see Mullennix *et al.*, *J. Acoust. Soc. Am.* **85**, 365-378 (1989); Martin *et al.*, *J. Exp. Psychol. Learn. Mem. Cog.* (in press)]. In the present study, the perceptual consequences of talker variability were investigated in further detail. In order to assess the relative magnitude of intratrial versus inter-trial changes in a talker's voice, a same-different reaction time judgment task was used. Performance was compared across three talker conditions: single talker, multiple talker across trials, and multiple talker within trials. In addition, the effects of talker variability over time were examined by observing performance across varying ISI delays. The results obtained provide further information about the nature of talker variability effects in the perception of spoken words and will be discussed in terms of their relationship to perceptual normalization processes in speech perception. [Work supported by NIH.]

4:34

YY15. Some perceptual correlates of phase in RELP coded speech. Margot Peet, Ward R. Evans, and Joseph Creekmore (The MITRE Corporation, 7525 Colshire Drive, McLean, VA 22102)

A series of experiments tested the contribution of phase information to the naturalness of residual excited linear prediction (RELP) coded speech, that is, how close a speaker's voice sounds under RELP coding to its unprocessed analog. Listeners were presented with a corpus of sentences uttered by eight speakers (four male and four female). Three classes of approximations of the residual phase spectra were employed on these sentences: quantized approximations for decreasing bit rates, noise-corrupted approximations for increasing noise levels, and combinations of these over varying window sizes. Listeners classified these utterances on the basis of their proximity to unprocessed speech. Results indicate that residual phase spectrum provides information about voice timbre and quality and acts as a cue to speaker identity in coded speech.

THURSDAY AFTERNOON, 25 MAY 1989

REGENCY C, 1:30 TO 5:00 P.M.

Session ZZ. Underwater Acoustics VII: Spatial and Temporal Variability of Ambient Noise II

Michael J. Buckingham, Chairman

Mission Management Department, Royal Aerospace Establishment, Farnborough, Hampshire GU14 6TD, England

Chairman's Introduction—1:30

Contributed Papers

1:35

ZZ1. Time-evolving characteristics of midfrequency ambient noise. W. S. Hodgkiss and F. H. Fisher (Marine Physical Laboratory, Scripps Institution of Oceanography, San Diego, CA 92152)

Measurements have been made of the ambient noise field between 25 and 300 Hz with vertical arrays at 32° N (124° W, 136° W, and 150° W). Due to substantial differences in weather at the stations, these measurements provide an opportunity to observe the effect of weather on the vertical distribution of ambient noise and how this distribution evolves with time. Array time series 3-4 h in length at each station have been analyzed. The time-evolving characteristics of ambient noise vertical directionality at each station will be compared. Specifically, the fluctuations of beam power for beams pointing near the horizontal will be compared with those pointing both upward and downward. [Work supported by ONR.]

1:50

ZZ2. Efficient simulation of the vertical structure of noise using the parabolic equation. R. B. Evans (SYNTEK, 1 Denison Avenue, Mystic, CT 06355), W. M. Carey, G. Botseas (Naval Underwater Systems Center, New London, CT 06320), and J. A. Davis (Planning Systems, Inc., New London, CT 06320)

The structure of noise at a vertical array in a range-dependent ocean basin can be simulated using the parabolic equation (PE). Noise sources

are distributed throughout the basin. The PE can be used to compute the field at the array due to each of the sources. The response of the array to the superposition of these noise sources is found by beamforming. The number of PE runs can be reduced by using the elements in the array as sources and relying on reciprocity, but even this indirect approach is impractical. An efficient and direct approach is to start at a distant noise source and superimpose additional noise sources on the PE field as the field is marched toward the array. Only one PE run is needed in a cylindrically symmetric ocean basin. The justification for this approach is discussed. An application demonstrates that noise at the horizontal can be caused by slope conversion of low-frequency noise sources over the basin boundaries. [Work sponsored by NUSC.]

2:05

ZZ3. Frequency-averaged, mean-squared coherence of simulated underwater explosive transients. Allen E. Leybourne (School of Engineering Technology, University of Southern Mississippi, Hattiesburg, MS 39406)

A method has been developed for determination of the frequency-averaged, mean-squared coherence (FMSC) of broadband transient acoustic signals, similar to underwater explosions. It has many of the properties of the more conventional mean-squared coherence (MSC), including a systematic bias, dependent upon the number of spectral values used in the estimate. The method has been demonstrated using a model of the underwater explosive direct arrival transient signal. As FMSC is aver-

aged over several frequencies the method compromises the functional knowledge of frequency dependence to achieve stable and relatively unbiased values. Results are compared to MSC, utilizing a calibration method developed during the study. The relative level of coherence between separate similar transient acoustic processes can be determined when the signals are precisely time aligned. FMSC can be used as the means of making this time alignment, thus making the necessary computations internally consistent. [Work supported by U.S. Navy, NORDA, Code 242, Arctic Acoustics.]

2:20

ZZ4. Low-frequency ambient noise measurements in the South Fiji Basin. R. Marrett and N. R. Chapman^{a)} (Defence Scientific Establishment, Auckland, New Zealand)

The effect of wind speed on low-frequency ambient noise has been measured in an experiment carried out in the South Fiji Basin. This site provides an ideal location for studying sea surface noise, owing to the low density of shipping in the basin. The data were recorded using a towed line array over an 11-day period in the winter storm season. The local wind speed was measured at the site, with an observed variation between 0–35 kn. The noise measurements in the frequency band 15–250 Hz were strongly correlated with the variations in the local wind speed. The relationship between wind speed v and noise level N is expressed by $N = B + 20 n \log(v)$. The constants B and n have been determined by fitting this model to the data at frequencies from 15–250 Hz. At 250 Hz, the values are 22 dB and 1.5 for B and n , respectively. This value for n is higher than the results of similar experiments carried out in the northern hemisphere oceans where the influence of shipping is greater. ^{a)} On exchange from Defence Research Establishment Pacific, FMO Victoria B.C. V0S 1B0, Canada.

2:35

ZZ5. Ambient noise measurements from 100 Hz to 80 kHz. Steven O. McConnell (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98105) and Michael P. Schilt (David Taylor Research Center, Ship Silencing Evaluation Division, Bremerton, WA 98314)

Measurements covering a broad frequency range from 100 Hz to 80 kHz have been made in Behm Canal, Alaska. This site represents a fairly deep embayment (400 m) with a very soft bottom (porosity of about 0.8) and, hence, the noise detected at the hydrophones is affected negligibly by multipath contributions except possibly at the very lowest frequencies. Data were gathered over a wide range of wind speeds (0–15 m/s) and during periods of rain and snow. Several unique and noteworthy results were obtained. Foremost was the observation that the wind-generated noise level measured during the winter was 5 dB lower than during the summer for the same wind speeds and air-sea temperature differences (air temperature about the same or colder than the sea surface). The summer data agree well with the most recent published measurements and are approximately 2 dB higher than the standard Knudsen/Wenz reference spectra. Most reported wind noise measurements are restricted to frequencies less than 20 kHz. Those that go beyond this frequency display a noticeable hump above the usual -17 dB/decade power-law slope, and the Behm Canal measurements show that this hump continues to 80 kHz, where the spectrum rejoins the extension of the canonical power-law slope. [Work supported by Battelle Pacific Northwest Laboratories and NORDA.]

2:50

ZZ6. Underwater noise generated by Columbia River hydroelectric dams. Robert T. Miyamoto, Steven O. McConnell (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98105), James J. Anderson, and Blake E. Feist (Fisheries Research Institute, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98105)

Low-frequency (10–1000 Hz) underwater noise measurements have been made in water within and upstream from four Columbia River hydroelectric dams. The motivation for these measurements was to map out the sound field within and upstream from the power dams as a first step in understanding the effect of this field on the behavior of migrating salmonids that must choose between the bypass system or intakes to the turbines. Eventually sound may be used to guide the juvenile fish safely past the turbine intakes and into the bypass system. Thus far, single hydrophone measurements have been made in the bypass slots within the dam and at a number of locations upstream from the dam. The noise level varies with location, decreasing as the hydrophone is moved upstream from the dam and as the hydrophone is moved closer to the water surface immediately upstream of the dam as well as in the bypass slot. The noise spectra below 200 Hz are highly modulated, displaying one or more sharp peaks, which indicates resonances in the structural generating mechanism or propagation path. The spectrum level and modulation vary significantly from one dam to another and sometimes from one configuration to another (e.g., when one of the turbines is on or off). A final set of measurements will be made at the Bonneville Dam using several hydrophones placed at a number of locations in the vicinity of the intake channel, and these may help identify sources and propagation paths to the hydrophone. [Work sponsored by U. S. Army Corps of Engineers.]

3:05

ZZ7. VLF acoustic ambient ocean noise measurements using Swallow floats. G. L. D'Spain, W. S. Hodgkiss, R. L. Culver, and G. L. Edmonds (Marine Physical Laboratory, Scripps Institution of Oceanography, San Diego, CA 92152-6400)

A set of independent, freely drifting, neutrally buoyant, VLF (1–20 Hz) acoustic sensors have been under development at the Marine Physical Laboratory. These Swallow floats each contain a three-component geophone for measuring VLF particle velocity, a magnetic compass, an 8-kHz acoustic localization system, and hardware to record 24 h of digitized data. By modeling the ambient noise field as a sum of independent, randomly distributed plane waves, the particle velocity power spectra estimated from geophone data can be converted into equivalent pressure power spectra. Derived pressure power spectra from four deep water (water depth > 3.5 km) deployments are compared to pressure spectra recorded by concurrently deployed sonobuoys and bottom-mounted hydrophones and to previously reported pressure spectral levels. In a recent deployment, four floats were equipped with VLF hydrophones as well as geophones. Comparisons are made between the power spectra estimated from geophone and hydrophone data collected at the same time and place by these floats. [Work supported by ONR.]

3:20

ZZ8. Bispectrum of ship-radiated noise. Melvin J. Hinich (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713), Davide Marandino (ELSAG S.p.A., Genova-Sestri, Italy), and Edmund J. Sullivan (Naval Underwater Systems Center, Code 01v, Newport, RI 02841)

The bispectrum of ship-radiated noise is estimated. The noise was received on a towed array being towed by the same ship that served as the noise source. The array was beamformed such that the forward end-fire beam pointed toward the towing platform and the broadside beam sampled the environment without the radiated noise of the ship. The results show that there exist frequency-dependent bispectral components in the ship's radiated noise, whereas the ambient noise does not contain any significant bispectral components. Since the existence of a nonzero frequency-dependent bispectrum indicates the existence of nonlinear com-

ponents in the noise-generating mechanism, it is concluded that the radiated noise of the towing platform contains such nonlinear mechanisms. Also, the bispectrum has the capability of indicating the presence of transients in the data. This property will be briefly discussed, and it will be shown that the data under analysis contain transients. Based on these results, it is concluded that the bispectrum could be used to indicate the existence of transient and nonlinear sources that would normally be hidden in the background noise when the usual spectral estimation procedures are applied.

3:35-5:00

Open Discussion

MODERATORS

Frederick H. Fisher

Scripps Institution of Oceanography, San Diego, California

Michael J. Buckingham

Royal Aerospace Establishment, Farnborough, England

THURSDAY AFTERNOON, 25 MAY 1989

COMSTOCK B, 2:00 TO 4:00 P.M.

Session AAA. Musical Acoustics VI and Speech Communication IX: Vocal Fold Function in Singing

Johan Sundberg, Chairman

Department of Speech Communication and Musical Acoustics, KTH (RIT), Box 70014, S-10044 Stockholm, Sweden

Invited Papers

2:00

AAA1. Relevant factors in modeling laryngeally controlled vocal vibrato. Thomas Shipp and E. Thomas Doherty (Speech Research Laboratory, VA Medical Center, San Francisco, CA 94121)

Rhythmic frequency oscillations that characterize vocal vibrato are produced by either of two mechanisms. One involves driving the system with respiratory pulses and the other by intrinsic laryngeal muscle activity. In this latter mode, vibrato is most readily produced when several laryngeal aspects are in order: (1) the singer's neuromuscular system has matured beyond a childhood level; (2) a critical balance of adductory-to-abductory force is applied to the vocal fold margins; and (3) the larynx is positioned vertically in the airway so that excessive vertical stiffening of the mucosal lining is avoided. When these conditions are met, the intrinsic laryngeal muscle activity to produce a given pitch changes from a relatively stable interference pattern to quasiperiodic modulated muscle activity oscillating at about 5.5 to 6.0 Hz. The average value of the modulated EMG pattern equals the steady-state EMG signal used to produce that frequency without vocal vibrato. Whether only the cricothyroid muscle or other intrinsic and/or extrinsic muscle are involved in this type of vibrato production is unknown at this time.

2:30

AAA2. Where has all the power gone? Ingo R. Titze (Voice Acoustics and Biomechanics Laboratory, Department of Speech Pathology and Audiology, The University of Iowa, Iowa City, IA 52242 and The Recording and Research Center, The Denver Center for the Performing Arts, Denver, CO 80204)

An upper limit on aerodynamic (pulmonary) power produced in shouting and singing is on the order of 1 W. The radiated acoustic power is less than 10% of this aerodynamic power, and in conversational speech can be as little as 0.1%. Major dissipative elements are the vocal folds, the lungs, the soft walls of the vocal tract, and turbulence in the airstream emerging from the glottis. Some analytical calculations and some computer simulation will focus on the distribution of losses along the vocal tract. In particular, difference between the singing mode and the speaking mode will be highlighted. Speculations will be made on the techniques by which a singer can optimize the system for maximum power output. [Work supported by NIH.]

AAA3. Acoustic studies of the voice source in male singers. Johan Sundberg (Department of Speech Communication and Music Acoustics, KTH (RIT), Box 70014, S-10044 Stockholm, Sweden)

This paper reviews various recent investigations with various co-workers of male singers' vocal fold function mainly studied by inverse filtering for deriving the waveform of the transglottal airflow during phonation. Theoretical and experimental studies confirmed the dominating acoustic importance of (1) the peak airflow \hat{U}_g for the amplitude of the voice source fundamental, and (2) the negative peak amplitude of the differentiated waveform dU_g/dt for the SPL of the radiated sound. Unlike nonsingers, some professional baritone and bass singers have been found to retain a high \hat{U}_g during loud tones at high pitches. Different strategies are possible for increasing dU_g/dt for loudness variation: increasing U_g , increasing the duration of the closed phase, and skewing the pulse shape. Different singers use different combinations of these alternatives depending on pitch, loudness range, and voice type. Some of the physiological correlates of these strategies can be postulated. Here, \hat{U}_g seems highly dependent on the degree of glottal adduction and dU_g/dt on subglottal pressure. However, influence of the breathing strategy has also been quantitatively demonstrated.

Contributed Papers

3:30

AAA4. Laryngeal adduction related to characteristics of the flow glottogram. Ronald C. Scherer (The Recording and Research Center, The Denver Center for the Performing Arts, 1245 Champa Street, Denver, CO 80204), Johan Sundberg (Department of Speech Communication and Music Acoustics, KTH, Box 70014, S-100 44, Stockholm, Sweden), and Ingo R. Titze (Voice Acoustics and Biomechanics Laboratory, Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52240 and The Recording and Research Center, The Denver Center for the Performing Arts, Denver, CO 80204)

Laryngeal adduction is a basic phonatory property related to voice quality and vocal pathology. In this study, simultaneous inverse filtered airflow and electroglottographic recordings were made for two male subjects phonating at deliberately varied pitches, loudness levels, and qualities. The goal of the study was to relate temporal characteristics associated with adduction with flow characteristics of the flow glottogram. A triangular model of laryngeal flow glottograms predicts that $\hat{U}_\omega/\hat{U} = k2\pi(Q_0 - Q_r)$, where \hat{U} is the peak flow, \hat{U}_ω is the negative peak flow derivative, Q_0 is the open quotient (the duration the flow is above the baseline, divided by the period), Q_r is the rise quotient (the duration the flow rises from baseline to peak, divided by the period), and $k = 1$. Empirical results suggest $k \approx 0.5$. These findings will be discussed, and a variety of adduction measures applied to the data will be compared.

3:45

AAA5. Vedic chanting and vowel intrinsic pitch: Evidence from an ancient source. T. V. Ananthapadmanabha, Kim Silverman (AT&T Bell Laboratories, Murray Hill, NJ 07974), and M. G. Prasad (Department of Mechanical Engineering, Stevens Institute of Technology, Castle Point, Hoboken, NJ 07030)

The purpose of this paper is to shed further light on the aetiology and speech specific nature of vowel intrinsic fundamental frequency (*IF0*), by investigating its presence in vedic chanting. This is one of the earliest forms of chanting (circa 5000 years old) which is still prevalent today, predating much of Indian classical music. It is interesting to study vedic chanting as it represents the borderline between speech and singing. There are three distinct tone levels in this chanting referred to as: (a) *udatta* high, (b) *anudatta* low, and (c) *svarita*, in-between. Sanskrit grammarians classify vowels according to the position of the tongue body and use the same labels as above for marking the tones. The interaction of vowel height and tone level in this vocal mode is investigated. Several vedic passages from three chanters with different vowels occurring with the same tone level and different tone levels for the same vowel have been recorded. Initial measurements of *F0* suggest that chanters can override the effect of *IF0* to achieve the same relative pitch values. This implies that the effect of *IF0* depends on vocal mode and an active control of *IF0* is possible.

Meeting of Accredited Standards Committee S1 on Acoustics

to be held jointly with

**Technical Advisory Group Meeting for ISO/TC 43 Acoustics,
and IEC/TC 29 Electroacoustics**

D. L. Johnson, Chairman S1

Larson-Davis Laboratories, 1681 West 820 North, Provo, Utah 84061

Standards Committee S1 on Acoustics. Working group chairs will report on their progress in the preparation of standards on methods of measurement and testing, and terminology, in physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Work in progress includes measurement of noise sources, noise dosimeters, integrating sound-level meters, and revision and extension of sound-level meter specifications. Open discussion of committee reports is encouraged.

The international activities in ISO/TC 43 Acoustics, and IEC/TC 29 Electroacoustics, will also be discussed. The Chairs of the respective U.S. Technical Advisory Groups for ISO/TC 43 (H.E. von Gierke), and IEC/TC 29 (V. Nedzelnitsky), will report on current activities of these Technical Committees.

THURSDAY AFTERNOON, 25 MAY 1989

REGENCY B, 3:00 TO 5:05 P.M.

Session BBB. Structural Acoustics and Vibration VII: Sound Radiation from Structures and Viscoelastic Materials

Allan D. Pierce, Chairman

Departments of Acoustics and Mechanical Engineering, Pennsylvania State University, University Park, Pennsylvania 16802

Chairman's Introduction—3:00

Contributed Papers

3:05

BBB1. Vibration transmission in and sound radiation from a scaled model of a ship's engine room. Alfred Levi and Richard H. Lyon (Departments of Ocean and Mechanical Engineering, MIT, Cambridge, MA 02139)

The vibration transmitted from a power plant foundation through supporting structure and fluid containing tanks into the hull structure, and then into the surrounding fluid has been studied for a scale model of a frigate ship. The scale model is approximately 1:6 and the scale frequency range is from 315–10 000 Hz. Transfer functions relating structural vibration and radiated sound power to input force (or injected mechanical power) have been measured. These experimental results have also been compared with SEA calculations of vibration transmission and sound radiation. Of special interest is the role of in-plane vibrations on the energy transmission within the structure, and into the sound field.

3:20

BBB2. Acoustic radiation from fluid-loaded structures with discontinuities. Charles Thompson and Rahul Sen (Electrical Engineering, University of Lowell, Lowell, MA 01854)

The presence of discontinuities such as joints, in a fluid-loaded structure results in farfield acoustic radiation even when subsonic wavenumbers prevail in the main body of the structure. The object of this study is to develop a rational model of this phenomenon using singular perturbation methods. When a flexural wave in a thin structure impinges on a joint, evanescent structural displacement fields are set up in the vicinity of the joint. This paper proposes to show that these fields are associated with rotation-dominated modes that are supported by shear-corrected plate theories of the Timoshenko–Mindlin type flexural mode. The method of matched asymptotic expansions will be used to obtain mode-conversion

coefficients, and it will be shown that the spatial wavenumber spectrum of the local field is instrumental in setting up an acoustic farfield in the surrounding fluid. An important mathematical issue is the proper description of evanescent fields; this analysis will be based on results obtained previously in connection with acoustic waveguides.

3:35

BBB3. Experimental facility for fluid/structure interaction research. Courtney B. Burroughs (Applied Research Laboratory, The Pennsylvania State University, P. O. Box 30, State College, PA 16804)

The boundary layer research facility (BLRF) at Penn State is being modified so that experimental research on the interaction between fluid boundary layers and vibration response of underlying structures can be conducted. Because the BLRF is a closed-loop glycerine tunnel, the boundary layer is thick, which provides space for detailed measurements required for research in fluid/structure interaction. In addition, laminar, transition, or turbulent flow can be generated and maintained. Modifications to the BLRF for fluid/structure interaction research include: (1) the introduction of an elastic and exchangeable structure in contact with the flow, (2) the reduction of background vibration to remove interference of structural response measurements by background vibrations, and (3) an increase in the measurement accessibility to the boundary layer. A description of the modified BLRF is presented along with technical support for several of the key design elements. Measurements to characterize and define the capabilities of the modified BLRF are discussed. [Work supported by M. Reischman, ONR Code 432F.]

3:50

BBB4. Vibratory response of heavily fluid-loaded plates to convecting pressure excitations via impulse response methods. D. D. Ebenezer and Peter R. Stepanishen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881-0814)

A general time domain approach is presented to evaluate the vibratory response of fluid-loaded plates to one-dimensional convecting pressure excitations. The velocity of the plate is expressed as a modal sum of the product of the *in vacuo* eigenfunctions and time-dependent coefficients. Previously developed [D. D. Ebenezer and P. R. Stepanishen, J. Acoust. Soc. Am. 82, 659-666 (1987)] sets of coupled convolution integral equations for the time-dependent modal velocity coefficients are rearranged to facilitate the numerical solution. In the previous approach, FFT methods were used to evaluate the uncoupled velocity responses. In the present approach, analytical expressions are obtained for the impulse responses of the modal admittances and the impulse responses of the modal radiation impedances are obtained via a combination of analytical and wave-vector-time domain techniques. The known terms in the coupled convolution integral equations are thus very easily obtained and a large number of modes can be included in the analysis. Numerical results are presented for the uncoupled and coupled velocity responses to loads moving across the plate. They clearly show that coupling between modes is significant and that the general characteristics of the vibratory response can be anticipated based on a knowledge of the Mach number and the fluid-loaded modal resonant frequencies. [Work supported by ONR.]

4:05

BBB5. Sound radiation of a finite plate with several types of boundary conditions. A. C. Berry, J. Nicolas (Groupe d'Acoustique de l'Université de Sherbrooke, Département de Génie Mécanique, Université de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada), and J. L. Guyader (Laboratoire de Vibration-Acoustique, INSA Lyon, 69621 Villeurbanne Cedex, France)

The sound radiation from baffled, thin, rectangular plates has generally been restricted to the simply supported plate analysis, for which simple relations hold. This paper investigates the influence on the acoustic radiation, of a more general type of boundary conditions, namely, edges presenting arbitrary translation and rotation stiffness. For such structures, the *in vacuo* modal basis is not known *a priori*. Thus the *in vacuo* vibrational response of the structure is determined using a polynomial expansion of the transverse displacement. In the limit of a low fluid-structure coupling (air loading), the *in vacuo* vibrational response may be used to calculate the acoustical perturbation generated in the fluid. Results are shown in terms of quadratic velocity of the plate, farfield acoustic pressure, acoustic power, and radiation efficiency of the plate. The influence of the boundary conditions is examined, and the limiting cases of simply supported, clamped, and free plates are presented.

4:20

BBB6. A simple apparatus for the measurement of the shear modulus of viscoelastic materials over 4 decades of frequency. Fred Schloss (Wilcoxon Research, 2096 Gaither Road, Rockville, MD 20850)

Two identical test samples are sandwiched between the metal ends of two small identical piezoelectric stacks, one acting as a transmitter having a displacement proportional to the applied voltage V , the other receiving a charge Q proportional to the force. Since the ends of the stacks are moving out of phase with zero motion at the center, no external fixtures are necessary. The modulus equals $Qt(VS)^{-1}(d_{33})^{-2}$, where t and S are the sample's thickness and cross-sectional area and d_{33} is a piezoelectric parameter. The useful frequency range is 1-10 000 Hz, which can be extended using smaller parts and different materials for the end plates, and the temperature range is 50-200 °C. The device is limited to low values of dynamic strain.

4:35

BBB7. Acoustic attenuation in syntactic foam and heavy silicone rubber composites. Raymond Lim and Roger H. Hackman (Naval Coastal Systems Center, Physical Acoustics Branch, Code 2120, Panama City, FL 32407-5000)

Recently, a particularly effective underwater sound attenuator has been described experimentally by workers at the Naval Surface Weapons Center using an impedance tube to measure echo returns [W. M. Madigosky *et al.*, *Multiple Scattering of Waves in Random Media and Random Rough Surfaces*, edited by V. V. Varadan and V. K. Varadan (Pennsylvania State University, 1985), p. 615]. The attenuator studied was composed of a rigid syntactic foam matrix loaded to 40% by volume with many large viscoelastic rubber inclusions. The present paper uncovers the fundamental attenuation mechanisms operating in this composite by analyzing the scattering behavior of a single spherical scatterer embedded in an elastic matrix. Effects due to resonant scattering by elastic and viscoelastic spheres are studied via the T -matrix formalism. The dynamics of the scattered field at the water-syntactic foam interface are also analyzed via classical elastic theory at a plane interface. The results of these analyses point to resonant compressional to shear mode conversion at the inclusion sites as ultimately responsible for the attenuations observed by Madigosky *et al.* Consequences of the mode conversion leading to attenuation include the trapping of backscattered shear waves and the enhancement of dissipation mechanisms within the inclusion.

4:50

BBB8. Boundary element analysis of viscoelastic structures involving fractional operators. Ke-jun Xie (Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27695-7910), Robert D. Ciskowski

(Numerically Intensive Computing Center, International Business Machine Corporation, Kingston, NY 12401), and Larry H. Royster (Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27695-7910)

The boundary element method (BEM) is applied to solve viscoelastic structural problems. The viscoelastic characteristic of the structure is represented using the fractional integrodifferential operators. The fractional operator representation of viscoelasticity provides a means to more accu-

ately describe the mechanical behavior of viscoelastic materials. In this paper, a direct BEM formulation for the transient dynamic analysis of three-dimensional viscoelastic structures modeled by the fractional operator constitutive equations is presented. The BEM formulation is developed in the Laplace transformed domain. The transient problem is first solved in the Laplace transformed domain and the time domain solutions are obtained by using an FFT based numerical inversion procedure. Example problems are solved using this BEM formulation and the results are compared.

Session CCC. Physical Acoustics IX: Inverse Problems in Acoustics I

Robert C. Waag, Chairman

*Department of Electrical Engineering, University of Rochester, Rochester, New York 14627***Chairman's Introduction—8:00*****Invited Papers*****8:05****CCC1. Inverse filtering for the analysis of vocal function.** Martin Rothenberg (Department of Electrical and Computer Engineering, Syracuse University, Syracuse, NY 13244)

The waveform of the volume velocity of the airflow through the larynx, as obtained by inverse filtering the volume velocity of the airflow exiting the mouth and nose, is a potential convenient noninvasive measure of the vibratory pattern of the vocal folds during voiced speech. Of special interest is a measure that is sufficiently robust for clinical use in the detection, monitoring, and treatment of voice disorders. The problems in this application are of three types: (1) The parameters of the inverse filter must be set to match the acoustic transfer function of the vocal tract; (2) glottal airflow is, in some way, not an accurate indicator of vocal fold motion; and (3) the analysis must be accurate under a wide range of normal and disordered voice qualities and possible nasal coupling. Because of these problems, it has been difficult to find a robust and accurate automatic procedure, even if the analysis is restricted to only the simplest case of open vowels. However, there appears to be a rather straightforward, although unconventional, solution to this inverse-filtering problem if (1) the vocal fold vibratory parameters of most interest are properly defined, (2) physiological restrictions on the possible vibratory patterns are taken into account, and (3) the nonlinear interaction between the glottal source and the vocal tract acoustics is used in the analysis procedure.

8:35**CCC2. Inverse problems in nondestructive evaluation.** R. B. Thompson (Department of Engineering Science and Mechanics and NDE Center, Iowa State University, Ames, IA 50011)

The goal of nondestructive evaluation is to obtain information about the state of a structural or electronic component that will allow a quantitative assessment of its future serviceability. Because of the ability to penetrate opaque media, ultrasound is one of the most widely used forms of interrogating energy. This paper will consider several current problem areas, ranging from the measurement of material properties such as porosity or grain structure through the characterization of the nature of discrete flaws and the determination of their size and shape. In each case, an interpretation of the measurement in terms of elastic wave scattering theory will be followed by a discussion of the inverse theories used to interpret the data, with emphasis on assumptions required to obtain unique solutions. The presentation will conclude with a discussion of future directions of needed research.

9:05**CCC3. Multiparameter inversion for acoustic and elastic media.** Robert Burridge and Gregory Beylkin (Schlumberger-Doll Research, Old Quarry Road, Ridgefield, CT 06877-4108)

Inversion and imaging of an acoustic two-parameter medium and for a three-parameter isotropic elastic medium from seismic data are discussed. The inverse problem is linearized by considering the actual medium as a perturbation from a known background model. By using the single-scattering approximation, integral equations relating the singly scattered field linearly to the unknown parameters are obtained. Asymptotic solutions of the linearized inverse problem are derived using the generalized Radon transform. This method is closely related to methods of seismic migration. Spatially varying background parameters and an almost arbitrary source-receiver configuration are allowed. The computation is performed in the time domain. All available data are used even if redundant. Combinations of the parameters which can be reconstructed best and worst are identified. The data may be read just once and in an arbitrary order.

CCC4. Inverse acoustic problems in the resonance scattering regime. G. C. Gaunaurd (Naval Surface Warfare Center, White Oak Laboratory, Research Department, Silver Spring, MD 20903-5000)

Solving an inverse scattering problem *here* means the extraction of physical (viz., composition and shape) information from the active echo returned to an underwater elastic target. This is the general area of *target identification*. In this sense, solving inverse scattering problems is relatively simple (!) in the low or high spectral regions because there a number of simplifying approximations may apply. However, for the intermediate or resonance region of all scatterers, the above approximations do not hold, and exact solutions or novel types of approximations are necessary, making the actual inversions of scattered waveforms *extremely* hard. A certain level of success has been achieved by exploiting the presence of certain resonance features that manifest themselves in the resonance region of the scattering cross sections (SCS) of submerged elastic targets. Examples of how and how well one can extract identifying sets of resonance features from an echo have been recently overviewed [G. Gaunaurd, J. Acoust. Soc. Am. Suppl. 1 **84**, S168 (1988)]. Although the main interest over the years has been the isolation of the resonance features in the SCS either by theoretical (viz., the background subtraction) and/or experimental (viz., samplings of echoes in their tail portions) means, little has been said about what to do with them, once isolated. This paper will show (idealized) examples illustrating how these resonance features can be actually used for unambiguous target characterization.

10:05

CCC5. Three-dimensional inverse scattering tomography using elastic waves. Steven A. Johnson (Department of Biomedical Engineering, University of Utah, Salt Lake City, UT 84112)

A 3-D ultrasound method that provides clinically significant (and diagnostically valuable) images of important tissue characteristics is described. This is made possible by an inverse-scattering technique that provides accurate, independent images of speed of sound, density, and absorption (with extension to five viscoelastic constants) at the very high spatial resolution of 0.5 to 1.0 wavelength of the incident ultrasound energy. Image size has increased from 16×16 pixels to 200×200 pixels and to $32 \times 32 \times 96$ voxels while maintaining accuracy and spatial resolution. No other methods, for example the *B*-scan mode of diffraction tomography based on the Rytov or Born approximations, have come close to matching this achievement. Further improvement in tissue imaging is possible by using the inverse scattering method in conjunction with a high performance reflection mode imaging method called synthetic focusing. The synthetic focusing method in nearly homogeneous medium produces medium to very large size, high spatial resolution images. The synthetic focus method also can be made to produce excellent images in the moderately high refracting environment of the human body by applying time shift and attenuation corrections obtained from medium resolution inverse scattering images of speed of sound and absorption. Regularization and constrained optimization methods have been used to improve the conditioning and therefore the speed of the inverse scattering methods.

10:35

CCC6. Exact analytic reconstruction in multidimensional inverse problems. Adrian I. Nachman (Department of Mathematics, University of Rochester, Rochester, NY 14627)

This presentation will review some recent mathematical breakthroughs in inverse scattering theory. Consider the pressure wave $p(\mathbf{r}, \mathbf{s}, \omega)$ generated by a point source \mathbf{s} oscillating harmonically with frequency ω :

$$\nabla_{\mathbf{r}} \cdot \{ [1/\rho(\mathbf{r})] \nabla_{\mathbf{r}} p(\mathbf{r}, \mathbf{s}, \omega) \} + \omega^2 K(\mathbf{r}) p(\mathbf{r}, \mathbf{s}, \omega) = -\delta(\mathbf{r} - \mathbf{s}).$$

The problem of determining the density $\rho(\mathbf{r})$ and the compressibility $K(\mathbf{r})$ from knowledge of $p(\mathbf{r}, \mathbf{s}, \omega)$ for sources \mathbf{s} and receivers \mathbf{r} located on a given surface S has been considered notoriously difficult. Traditional approaches rely on Born or Rytov approximations. A new constructive method for recovering ρ and K from measurements on a surface S with arbitrary geometry, at two frequencies ω_1, ω_2 , will be presented. The procedure involves the solution of certain integral equations on S . No simplifying approximations are made, and the reconstruction is, in principle, exact.

Contributed Papers

11:05

CCC7. Scattering inversion using the ramp response. Bill D. Cook (Cullen College of Engineering, University of Houston, Houston, TX 77204-4792)

The objective of inversion is to have a result that is interpretable. Impulse and frequency techniques suggest that all could be known if one

had the privilege of using all frequencies and could observe the scattering from a multitude of directions. Realistically, one has a limited bandwidth of frequencies and often only views from a few directions. At the sacrifice of inversion detail, one can use the ramp response that requires a much smaller bandwidth. Shape, size, and volume information can still be extracted from a few views. Experimental results at audible and ultrasonic frequencies demonstrate the viability of this technique.

11:20

CCC8. Phase space and path integral methods in direct and inverse scattering. Louis Fishman (Department of Mathematics, Colorado School of Mines, Golden, CO 80401)

Direct and inverse wave propagation models in ocean seismoacoustics are often complicated by the extended, variable, three-dimensional channeling environments encountered. Even for the zeroth-order picture of a transversely inhomogeneous environment, the domain of interest is entirely within the scattering regime, with the subsequent absence of an "asymptotically free," or homogeneous, region. While there are numerous methods for attacking the one-way direct calculation, most approaches to the corresponding inverse (tomographic) problems essentially rely on a linearization about a well-chosen, zeroth-order model. Phase space and path integral techniques, which extend homogeneous Fourier methods to inhomogeneous environments, lead to comprehensive mathematical and computational direct models, in addition to providing the framework for the exact solution of the transversely inhomogeneous refractive index profile reconstruction problem. These pseudodifferential and Fourier integral operator methods are largely based upon the properties and interrelationships of the corresponding operator symbols. This talk will present the most recent developments and applications of the phase space and path integral methods, as applied to the scalar Helmholtz equations, for one-way direct and inverse modeling. [Work supported by NSF, AFOSR, ONR.]

11:35

CCC9. Inverse problem for an immersed solid/liquid/solid plane structure. J. L. Izicki, O. Lenoir, P. Rembert, G. Maze, and J. Ripoché (Laboratoire d'Electronique et d'Automatique Ultrasons, Université Le Havre, Place Robert Schuman, 76610 Le Havre, France)

The scattering from an immersed plane multilayer is studied in order to determine the physical parameters of the multilayer. It is composed by two solids that are coupled by a film of water about 0.3 mm thick. The

insonification is performed by short pulses. The frequency range is about from 0.5 to 3 MHz. The longitudinal phase velocities of the two solids are obtained through an angular study. The FFT processing of parts of the signal is performed at normal incidence. The thicknesses of the two solids are thus derived. Absolute measurements of the specular echo give the density of the first solid. The density of the second solid is obtained from an absolute measurement of the reflection coefficient of the whole structure. At oblique incidence, at resonances, guided waves are generated. Their spectral analysis allows the shear phase velocity of the two solids to be determined. All parameters are determined within an error of 10%. [Work supported by DRET DGA Contract 87-261.]

11:50

CCC10. Theoretical development of acoustic agglomeration of aerosol particles. Limin Song (Noise Control Laboratory, The Pennsylvania State University, University Park, PA 16802)

The orthokinetic interaction is recognized as a major mechanism responsible for rapid agglomerations of polydisperse aerosols in a sound field. The acoustic agglomeration theory based on this mechanism was developed by Mednikov in the 1960s. However, this theory has two basic problems that are not answered satisfactorily: How many small particles in an agglomeration volume created by a large particle will actually be collected in one cycle and how many small particles outside will be attracted into the volume so that agglomeration can continue for successive cycles? These problems have been solved to the first-order approximation in this investigation. A general expression for the specific acoustic agglomeration rate is derived by a statistical approach. The efficiency of collision between particles in a sinusoidal wave field is given explicitly. In addition, the analysis of the relative motion between particles was performed by including the effects of acoustic scattering from particles. The results show that a small particle is strongly attracted toward a larger particle once they cross over each other. This transverse attraction provides an efficient mechanism for the fill-up of the agglomeration volume.

FRIDAY MORNING, 26 MAY 1989

NEWHOUSE II, ROOM 254, 8:15 TO 11:29 A.M.

Session DDD. Speech Communication X: Perception and Analysis of Consonants

Cynthia N. Connine, Chairman

Department of Psychology, SUNY at Binghamton, Binghamton, New York 13901

Contributed Papers

8:15

DDD1. The role of rate of transition in the perception of place of articulation. Renee A. E. Zakia and John Kingston (DMLL, Morrill Hall, Cornell University, Ithaca, NY 14853-4701)

Listeners can reliably identify the place of articulation of bilabial and alveolar stops in severely truncated CV syllables from just the initial 20 ms, but they need 40 ms or more to identify a velar. The need for a longer interval may reflect listeners' expectations that rate of transition should be slower for velars because of the slower opening gesture at this place of articulation [Kewley-Port *et al.*, *J. Acoust. Soc. Am.* **73**, 1779-1783 (1983)]. This paper investigates the perceptual effects of differences in rate of transition in two experiments. First, synthetic CV syllables, where the direction and magnitude of the F_1 , F_2 , and F_3 transitions matched

natural [ba,da,ga] syllables, but where the transition durations were varied in 15-ms steps between 15 and 120 ms were presented to listeners for two-interval, forced-choice discrimination and for identification along a stop-glide dimension. If listeners expect longer transitions with velars, then they should accept stimuli with longer transitions as stops when the formants indicate a velar than when they indicate a bilabial or alveolar. Second, variation in F_2 and F_3 transitions between values appropriate for [da] and values appropriate for [ga] were combined with variation in the rate of transition; rate of transition was varied in 5-ms steps between 20 and 60 ms. These stimuli were also presented to listeners in two-interval, forced-choice discrimination and identification of place of articulation tasks. It is expected that faster rates of transition will shift listeners' response to stimuli with F_2 and F_3 transitions near the [d]/[g] boundary toward [d] and slower transitions will shift them toward [g].

DDD2. Effects of first formant onset qualities on VOT judgments can be explained by auditory processes not specific to humans. Keith R. Kluender (Department of Psychology, University of Wisconsin, Madison, WI 53706)

Both *F1* transition duration and *F1* onset frequency have been proposed to be perceptually significant in categorization of voiced and voiceless syllable-initial stops. Transition duration *per se* may not, however, explain the fact that, for longer transitions, longer *F1* cutback is required in order to perceive a stop as voiceless. Longer transitions result in lower *F1* onsets at any duration of cutback greater than zero, and it is possible that the major effect of *F1* is determined by its frequency at onset. In this study, *F1* transition duration, onset frequency, and slope were varied across four types of *F1* transition in which one of the three variables (onset frequency, duration, slope) was held constant while the other two were allowed to vary. Each of the four *F1* types was used in syllables with higher formants appropriate for labial, alveolar, and velar places of articulation. By far, the best predictor of identification of these stimuli by human listeners was *F1* onset frequency; *F1* duration, *F1* slope, and place of articulation had little or no effect on labeling boundaries. Six Japanese quail (*Coturnix coturnix japonica*), trained to respond differentially to voiced versus voiceless stops, evidenced "labeling" behavior almost identical to humans' with the same stimuli. These results are taken to provide strong evidence that *F1* onset frequency is the primary determinant of shifts in VOT boundaries across place of articulation, and that an auditory mechanism not specific to humans is responsible.

8:39

DDD3. The perception of stop consonants: Spectral tilt revisited. Kevin H. Richardson and James R. Sawusch (Department of Psychology, SUNY at Buffalo, Amherst, NY 14260)

The claim that invariant features involving spectral tilt, as proposed by Lahiri *et al.* (1984), are used by human listeners when classifying stop consonants was explored. Subjects were presented with stop-vowel syllables representing [b], [d], and [g] with a range of vowels in an identification task. Stimuli were either natural, synthetic (based on the natural set), or modified tokens (synthetic tokens containing a burst which, according to the Lahiri *et al.* metric, should be misidentified). Results indicate that the change in spectral tilt plays little role in stop perception. Only [g] showed any evidence of a systematic effect. To investigate further the difference found in the identification task, tone analogs mimicking the [b], [d], and [g] CV sets were produced. Both speech and nonspeech groups were run to examine the perceptual locus of any effects of change in spectral tilt. The results of these two groups will be compared to those of the speech groups in the first experiment. The implications for theories of auditory to phonetic coding of stops and possible invariant attributes in stop perception will be discussed.

8:51

DDD4. Spectral shape factors for speaker-independent automatic recognition of stop consonants. Zaki B. Nossair and Stephen A. Zahorian (Department of Electrical and Computer Engineering, Old Dominion University, Norfolk, VA 23529)

A series of automatic recognition experiments was conducted with naturally produced English stop consonants /b,p,d,t,g,k/ in syllable initial position. The objectives of the experiments were to investigate in detail the effectiveness of spectral shape factors, as both dynamic and static features for automatic recognition of stop consonants. Spectral shape factors were computed as the discrete cosine transform coefficients

(DCTCs) of the magnitude spectra. The database used in these experiments consisted of 2481 CVC syllables spoken in isolation by ten males, ten females, and ten children. In all experiments, 15 speakers were used to train the classifier and the other 15 speakers were used for evaluation. For the case of dynamic features, DCTCs were computed over a 50-ms interval beginning with the burst using 7-ms frames spaced every 5 ms. For the static case, the DCTCs were computed from one 25.6-ms frame beginning at the burst. Automatic classification results, based on the test data, were 87.2% for the dynamic spectral shape features versus 60.1% for the static case. Dynamic features timed to begin with the formant transition area resulted in 44.6% recognition rates. Control experiments with formants resulted in much lower recognition rates under every condition tested. In summary, these results are in agreement with the theory that dynamic spectral shape, spanning an interval of approximately 50 ms beginning at the burst, carries most of the information for initial stop consonants [D. Kewley-Port, *J. Acoust. Soc. Am.* 73, 322-335 (1983)]. [Work supported by NSF.]

9:03

DDD5. Effects of speaking rate are moderated by phonological structure. Lydia E. Volaitis (Department of Psychology, Northeastern University, Boston, MA 02115)

Changes in speaking rate can alter the duration of properties that convey segmental information. For example, with an increase in syllable duration, the distribution of VOT values for labial stops, especially the voiceless /p/, shifts upward along the VOT continuum. Perceptual adjustment to this rate effect entails a shift in the category boundary toward longer VOT values for longer syllables and, as recently reported, a change in which stimuli along the VOT continuum are perceived to be best category exemplars [J. L. Miller and L. E. Volaitis, *J. Acoust. Soc. Am. Suppl.* 1 80, S125 (1986)]. The present two-part investigation focuses on whether the location of best exemplars is solely a function of syllable duration, or depends as well on the phonological structure of the syllable, specifically, the place of articulation of the initial consonant. Results of the production study have confirmed that as place moves from labial to alveolar to velar, VOT increases, and have shown that this occurs across a range of syllable durations. The parallel perception study, currently in progress, addresses whether at any given syllable duration, the location of the best exemplars varies with place, in accord with the production data. Such a result would indicate that the location of the best exemplars is a function of the phonological structure of the syllable, in conjunction with its duration. [Work supported by NIH.]

9:15

DDD6. The effect of position of stress on the durations of stop consonants in American English. Alice E. Turk (DMLL, Morrill Hall, Cornell University, Ithaca, NY 14853-4701)

It has been documented for American English that /t/ and /d/ are realized as very short alveolars (flaps) following a stressed vowel nonfinal initially: [\check{V}_V (...)] or [$\check{V}CV_V$ (...)] [Zue and Laferrière (1979)]. This study investigates the duration of other stop consonants in the flapping environment to see whether the shortening of /t/ and /d/ is a phonologization of a more general phonetic principle. Stathopoulos and Weismer (1983) report that in nonsense words, stop closure durations for all places of articulation are shorter in the post-stress environment. In a study of two speakers using natural speech, one speaker's consonants shortened significantly ($p < 0.05$) in the flapping environment. However, the fact that the second speaker did not exhibit the same shortening pattern suggests the need to expand the number of speakers; results from such a larger subject pool will be reported.

DDD7. Fricative perception: Frication and transition cues. Fan-Gang Zeng and Christopher W. Turner (Communication Sciences and Disorders, Syracuse University, Syracuse, NY 13244-2280)

By presenting fricative-vowel tokens across a range of intensities and determining the audibility of various portions of the tokens, the present study investigated the cues for fricative perception. The frication cue was defined as gross spectral-shape differences residing in the steady-state frication noise burst; while the transition cue was defined as dynamic formant changes in the transition portion between the fricative and its adjacent vowel. Both natural and synthetic syllables, with duration and amplitude normalized, were used in closed-set recognition tasks by four normal-hearing subjects. The presentation levels varied from 0%–100% audibility for individual fricatives. Results suggest that the frication burst may be a sufficient cue for the correct perception of fricatives; while the transition cue, although it may not be necessary, plays an important role in helping to identify the place information, particularly at low presentation levels where the frication cue is always not audible. Two other experiments, using truncated stimuli, with frication or transition cues presented independently, further supported our conclusions. [Work supported by Syracuse University Senate Research Grant and Deafness Research Foundation.]

DDD8. Integrality in consonant perception: Vowel environment influences consonant perception. Gail R. Tomiak (Department of Psychology, State University of New York at Buffalo, Amherst, NY 14260)

Previous research designed to examine the nature of phonetic processing through the use of noise-tone analogs to speech syllables has demonstrated that the phonetic coding of noise and tone segments as consonant and vowel within a syllable proceeds in a mutually dependent (integral) fashion. However, the auditory coding of these segments is independent (separable). The purpose of the present experiment was to determine whether consonant information is also processed in an integral fashion with respect to vowel information contained in an adjacent syllable. Tone–noise–tone analogs of V_CV disyllables were used in a speeded classification task. The results indicate that the phonetic coding of the noise as a fricative is integral with respect to both the preceding and following vowel. Thus the perceptual coding of consonants seems to proceed with reference to a surrounding vowel environment regardless of syllable boundaries. The implications of these results for the phonetic coding of speech, and recent proposals regarding the priority of vowel information in perception, will be discussed. [Work supported by NINCDS.]

9:51–10:05

Break

10:05

DDD9. Phone restoration. Bruno H. Repp (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695)

When a speech segment is replaced by an extraneous sound of not too dissimilar spectral structure, the speech often seems intact (the “phoneme restoration” effect). Does this occur because the missing phonological segment is filled in at the level of lexical access, or because the missing phonetic information is filled in at a prelexical stage? This issue was investigated by probing subjects’ perception of the extraneous noise burst that replaced the frication of an [s] inside a short sentence. The results showed that the noise was perceived as lower in “pitch” than an identical noise presented before or after the sentence, or inserted into the silent closure of a [t^h] in a similar sentence. This suggests that the [s] was restored phonetically by auditory subtraction from the noise occurring in its place, which left a perceived extraneous residue that was depleted of high frequency energy. Additional experiments investigated whether this “phone restoration” was induced by phonetic cues to the missing segment in adjacent signal portions, or whether lexical access played a role. [Work supported by NICHD.]

10:17

DDD10. Identification of place of articulation in nasal phonemes using a time-frequency approach. Edgar F. Velez, Richard G. Absher (Department of Electrical Engineering, University of Vermont, Burlington, VT 05405), and James F. Lubker (Department of Communication Science and Disorders, University of Vermont, Burlington, VT 05405)

Acoustic cues to place of articulation are present in the highly nonstationary region between nasal consonants and vowels. A need for a speech processing tool capable of accurately representing these transient events led to consideration of the Wigner–Ville distribution (WVD). Previous use in speech analysis was limited due to the presence of multiple interference terms (crossterms), which hindered its direct interpretation. These

crossterms can be attenuated by time-frequency smoothing (SWVD). Time and frequency resolution may be chosen independently, thus avoiding the trade-off existing in wide- and narrow-band spectrograms. Despite its apparent complexity, the SWVD can be computed about four times faster than a spectrogram. SWVD analysis of nasal-vowel transitions accurately shows fine spectrotemporal details in the murmur release, initial slope of the formant transitions, and abrupt energy changes. Identification of place of articulation experiments performed on labial and alveolar nasal consonants in various vocalic contexts in continuous speech, using the SWVD of 20- to 60-ms segments around the transition, shows good correlation with previous perceptual experiments.

10:29

DDD11. Training Japanese listeners to identify /r/ and /l/. John S. Logan and Scott E. Lively (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Native speakers of Japanese learning English generally have difficulty differentiating the phonemes /r/ and /l/, even after years of experience with English. Previous research that attempted to train Japanese listeners to distinguish this contrast using synthetic stimuli showed little success, especially when transfer to natural tokens containing /r/ and /l/ was tested. In the present study, a procedure that differed from these earlier attempts was used. Japanese subjects were trained in an identification paradigm using as stimuli multiple natural exemplars contrasting /r/ and /l/ from a variety of phonetic environments. A pretest–posttest design combined with a test of generalization containing novel natural tokens was used to assess the effectiveness of training. Analysis of pilot data from a small group of subjects showed that the new procedure was more robust than earlier procedures. The results demonstrate the importance of stimulus variability and task-related factors in training second language learners to perceive novel phonemic contrasts that are not distinctive in their native language. [Work supported by NIH.]

DDD12. An audio-visual investigation of linguistic stress perception. Lawrence D. Rosenblum (Department of Psychology, Wellesley College, Wellesley, MA 02181)

The true acoustic correlate for linguistic stress has not yet been found. While it was originally thought that listeners base stress judgments on syllable intensity, it has been shown that intensity, fundamental frequency, duration, and spectral structure can all act as effective information for stress perception [D. Isenberg and T. Gay, *J. Acoust. Soc. Am. Suppl.* 1 63, S21 (1963)]. Because no single simple acoustic dimension seems to be dominant, it has been proposed that listeners base stress judgments on an articulatory property such as vocal effort. Since such properties can be specified optically as well as acoustically, a study to test this hypothesis was conducted which used conflicting audio-visual presentations of a speaker producing tokens from two noun-verb pairs (CONvict-con-VICT and PERmit-perMIT). The prediction was made that if stress judgments are based on perception of articulatory dynamics rather than on simple acoustic parameters, then judgments should be affected by visual as well as auditory information. It is shown that stress judgments are affected by visual information even when subjects (1) are instructed to base their judgments on only what they hear and (2) cannot detect a discrepancy between the audio and visual components. Similar results are also shown for noun-verb tokens distinguished by an auditory dimension that cannot be specified visually (fundamental frequency) indicating that a more general articulatory property, such as vocal effort, might be the basis of stress judgments.

10:53

DDD13. Effects of lip-read information on auditory perception of Japanese syllables. Kaoru Sekiyama and Yoh'ichi Tohkura (ATR Auditory and Visual Perception Research Laboratories, Inuidani, Seikacho, Kyoto 619-02, Japan)

The McGurk effect is a phenomenon that demonstrates a perceptual fusion between auditory and visual (lip-read) information in speech perception under visual-auditory discrepancy condition (using dubbed video tapes). This paper examined the relation between the "McGurk effects" and the intelligibility of auditory stimuli. A female narrator's speech was video taped for ten Japanese syllables (/ba/, /pa/, /ma/, /wa/, /da/, /ta/, /na/, /ra/, /ga/, /ka/). The video and audio signals for these ten syllables were combined, resulting in 100 audio-visual stimuli. These stimuli were presented to ten subjects who were required to identify the stimuli as heard speech in both noisy and noise-free conditions. For both conditions, the intelligibility of the auditory stimuli was measured, presenting the auditory stimuli alone. In the noise-free condition, the McGurk effect was small and found only in conditions in which the intelligibility of the auditory stimuli was not 100%. In the noisy condition, the

McGurk effect was very strong and widespread. These results suggest that incomplete intelligibility of auditory stimuli is necessary for the McGurk effect.

11:05

DDD14. Larynx frequency information for lipreading speech in Mandarin Chinese. Yuk Ching Ching (Department of English, Chinese University of Hong Kong, Shatin, N.T., Hong Kong)

Prosodic contrasts are important sources of information in a tone language like Chinese. Each syllable is characterized by a tonal contour that determines the lexical meaning of the word in isolation. In continuous speech, the overall stress and intonation patterns contribute significantly to understanding speech. These contrastive speech patterns are directly related to the fundamental frequency of the speaker's voice, which is relatively invisible. When speech communication is dependent on lipreading, auditory voice pitch information can thus be the most effective aid. That this is so in an English-speaking context has been shown in numerous studies. The present work evaluates the contribution of larynx frequency information to lipreading connected discourse, and specifically to resolving ambiguities that are primarily signaled by contrastive stress patterns and tone patterns in Mandarin Chinese. The results have important implications for the design of speech processing aids and the development of speech perceptual assessments in a tone language community.

11:17

DDD15. Interpreting visual speech signals using neural networks. Ben P. Yuhas (Speech Processing Laboratory, Department of ECE, Johns Hopkins University, Baltimore, MD 21218)

When the acoustic speech signal is degraded by noise, compensatory information can often be obtained by watching the speaker's mouth [W. H. Sumby and I. Pollack, *J. Acoust. Soc. Am.* 26, 212-215 (1954)]. The aim of the work to be presented is to explore the structure of this compensatory information, as it might be used to assist automatic speech recognition. Starting with visual images taken from laser disc recordings of speaker's faces [Bernstein *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 82, S22 (1987)], a small box centered at the mouth is identified and extracted. A variety of neural networks are then taught to process these images based upon two interpretation schemes. The first approach is to classify the visual images as belonging to a particular class. The second approach is to extract acoustic constraints directly from the image. Toward this end, neural networks are trained to estimate the short-term power spectral envelope of the acoustic signal given the corresponding visual image as input. Performance is evaluated on nine vowels obtained from two speakers. These results will be reported and compared with results from more traditional classification and estimation techniques. [Work supported by AFOSR Contract No. 86-0246.]

Session EEE. Engineering Acoustics VIII: Scattering from Acoustic Materials

Caroline L. Fu, Chairman

*Research Engineering Division, Boeing Aerospace Company, P. O. Box 3999, Seattle, Washington 98124***Contributed Papers**

8:30

EEE1. Refraction and reflection of an obliquely incident finite amplitude plane P wave at a plane solid-solid interface. Kun-Tien Shu and Jerry H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

An earlier paper [K. T. Shu and J. H. Ginsberg, *J. Acoust. Soc. Am. Suppl. 1* **84**, S5 (1988)] proved that the significant finite amplitude aspects of mode conversion of a plane dilatational wave obliquely incident at a plane stress-free boundary may be described in terms of Earnshaw-type solutions for the dilatational and vertically polarized shear waves. In the work described here, the method is extended to an obliquely incident finite amplitude plane dilatational wave at a plane interface between two bounded isotropic solids. It is shown that, not as a consequence of nonlinear interaction effects, but rather as a consequence of nonlinear self-action effects, cumulative growth of higher harmonics occurs in the incident, reflected, and refracted waves. A number of special circumstances of wave reflection-refraction are examined, relative to the incidence angle and the material properties of the two media. Also given are the deformed incident, reflected, and refracted waveforms near one shock formation distance. [Work supported by NSF and ONR.]

8:45

EEE2. Experimental validation of a model for predicting the impedance of multilayer systems by the transfer matrix method. R. Woodcock and M. Hodgson (Groupe d'Acoustique, Département de Génie Mécanique, Université de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada)

As has been demonstrated by various authors, transmission line theory is a powerful technique for studying the sound transmission of multilayer systems with infinite lateral extent. This model has been extended to the study of the surface impedance of such systems. It is shown how the transfer matrices are derived from fundamental acoustic equations. These equations contain information on the conventions adopted concerning the assumed time dependence and orientation axis. It is shown that it is convenient to account for this during the juxtaposition of the quadrupoles in cascade, this modeling the superposition of the various layers. An experimental validation of this model has been undertaken on two types of mineral wool, placed in different multilayer configurations in an impedance tube, using the Chung and Blaser transfer function technique. The measured impedances are in excellent agreement with those predicted using the Delany and Bazley impedance model. Finally, it is shown that the model is only valid if each layer of the system is homogeneous.

9:00

EEE3. Reflection-reduction model for a viscoelastically coated plate. H. C. Strifors (Swedish Defense Research Establishment, S-10254 Stockholm, Sweden) and G. C. Gaunaud (Naval Surface Warfare Center (R43), White Oak, Silver Spring, MD 20903-5000)

A study is presented of the reflection of a plane cw sound waveform incident on a fluid-loaded elastic plate, quantitatively describing how and how much this reflection is selectively reduced by covering the plate with an ideally bonded viscoelastic layer. The exact equations of acoustics, elastodynamics, and viscoelastodynamics (in the Kelvin-Voigt model) are used to describe the behavior of the external fluid, the elastic plate, and the absorbing coating, respectively. By means of the principle of fading memory and a recently developed formulation [Strifors and Gaunaud, *J. Acoust. Soc. Am.* **85**, 995-1004 (1989)] with an appropriate number of time constants, the solution is generated in all three media. Once these fields are determined, the reflection (or transmission) coefficient for the coated structure is easily constructed. A number of calculated plots for the echo-reduction levels and their associated phase shifts versus frequency are displayed to illustrate the versatility of this predictive approach in a variety of situations. The approach can be used to evaluate the reflection-reduction capacity of the absorbing layer.

9:15

EEE4. Forced vibration of a submerged spheroidal shell using variational principles. Pei-Tai Chen and Jerry H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Variational principles are employed to study acoustic radiation from a submerged prolate confocal spheroidal shell. Structural vibration is described by the method of assumed modes, based on Hamilton's principle, in which bending and membrane stresses are included in the strain energy. The surface pressure is described by a variational principle that is derived from the surface Kirchhoff-Helmholtz integral equation. This approach was employed earlier [*J. Acoust. Soc. Am. Suppl. 1* **84**, S6 (1988)] to study a spherical shell under a harmonic, concentrated force. A singularity that occurs in the set of coefficients describing the coupling between surface displacement and pressure was found to require further attention. In the case of a sphere, evaluation of the contribution of this singularity, which arises in an integration over the surface, is expedited by the fact that the singularity is independent of position of the field point relative to the source point. The present paper addresses the evaluation of the coupling coefficients for an arbitrary body of revolution. Convergence is illustrated numerically, and results are compared with those obtained by Yen and Dimaggio from their analytical solution based on membrane theory [*J. Acoust. Soc. Am.* **41**, 618-626 (1967)]. [Work supported by the Office of Naval Research, Code 1132-SM.]

9:30

EEE5. Application of the finite element method to the modeling of plane wave scattering from compliant tube gratings at oblique incidence. A. C. Hennion, R. Bossut, J. N. Decarpigny (Institut Supérieur d'Electronique du Nord, 41 Boulevard Vauban, 59046 Lille Cedex, France), and C. Audoly (Groupe d'Etude et de Recherche de Détection Sous-Marine, Le Brusc, 83140 Six Fours les plages, France)

The modeling of compliant tube gratings has been performed with the help of the finite element method, using the ATELA code [Decarpigny *et*

al., J. Acoust. Soc. Am. **78**, 1499–1507 (1985)]. To do this, only the unit cell of the periodic structure, including a small part of the surrounding fluid domain, has to be meshed, due to the use of a classical Bloch-type relation between the displacement components of points that are separated by the grating spacing. Then, the effect of the external fluid domain is accounted for by matching the pressure field in the finite element domain to simple ingoing and outgoing plane-wave expressions. This paper describes results obtained for the scattering of a plane wave from different compliant tube gratings, including internal losses, at oblique incidence. First, it shows that a nice agreement has been found between predictions from an analytical model and finite element results for the insertion loss of an infinite elastic plate, thus validating the method. Then it describes the case of cylindrical tube gratings embedded or not in a viscoelastic layer, and shows the effects of low-stiffness encapsulant on insertion loss curves. Finally, double-layer gratings with different separations are also considered.

9:45

EEE6. Modeling of acoustic environment for the purpose of noise cancellation. Osama A. Mohammed, Kang Yen, Vijay Raman, and Kurt Ramdin (Department of Electrical Engineering, Florida International University, Miami, FL 33199)

The specific noise-cancellation problem being considered here is the active cancellation of acoustic noise by the use of electronic earphones. Since it is important to achieve cancellation of ambient noise only, and not of warning sounds and speech, an adequate understanding of the noise environment in which the earphones are to be used is required. A key step in the approach taken by these authors is to initially extract information regarding specific noise environments off-line. This information can then be used to define details of control structure and values for the canceler in that environment. During actual usage of the earphones, the algorithm must adapt to fine-tune itself to the environment at that time. Recordings

of ambient noise were made for specific environments. Data-analysis programs were developed and used to construct a model of the stationary noise environment and extract the relevant parameters. These are used to define the structure, parameter range, and computational requirements of the noise canceler. [Work supported by the Florida High Technology and Industry Council.]

10:00

EEE7. Low-frequency transmission loss by alternate resonance tuning (ART). Donald B. Bliss and James A. Gottwald (Department of Mechanical Engineering and Materials Science, Duke University, Durham, NC 27706)

Achieving adequate low-frequency transmission loss through lightweight flexible paneled walls is an important problem, particularly for interior noise reduction of propeller aircraft. Alternate resonance tuning (ART) is a new method of blocking sound transmission through such structures by controlling the dynamics of the wall panels. Adjacent acoustically compact panels are designed to have resonant frequencies alternately above and below the frequency to be canceled. At the design frequency, such panels oscillate with equal amplitude but nearly out of phase. The resulting cancellation from acoustic sources on adjacent panels causes cutoff of the transmitted acoustic modes, which become evanescent and nonpropagating. The concept differs from other noise reduction methods in that it does not rely on reducing the wall motion, but rather alters the wall wavenumber spectrum. There is no serious weight penalty associated with the method, and increasing the structural damping is not advantageous. Experimental and analytical results are presented to demonstrate the effectiveness of the ART concept. Experiments show that very large transmission losses with reasonable bandwidth can be achieved by a simple ART panel barrier in a duct. These findings are corroborated theoretically by an acoustic branch analysis, and results of a parametric study are presented to show the effect of varying system parameters.

FRIDAY MORNING, 26 MAY 1989

REGENCY B, 8:30 TO 11:30 A.M.

Session FFF. Musical Acoustics VII, Psychological Acoustics VIII, and Physiological Acoustics VI: Musical Dynamics

William M. Hartmann, Cochairman

Department of Physics, Michigan State University, East Lansing, Michigan 48824

Caroline Palmer, Cochairman

Department of Psychology, Ohio State University, Columbus, Ohio 43210

Invited Papers

8:30

FFF1. Unmasked and partially masked loudness in musical dynamics. Eberhard Zwicker (Institute of Electroacoustics, Technical University München, 8000 München 2, Federal Republic of Germany)

In music, two varieties of loudness are important. In one case, the loudness of the whole orchestra is being judged. The sensation may be characterized by the expression, "The finale was impressively loud." In the second case, the loudness of a single voice or instrument that is almost totally masked by other instruments is being judged. Here, the sensation may result in a statement such as, "The voice of the singer was hardly audible." There are two reasons for such masking, both realized in music: the spectral (simultaneous) partial masking and the temporal (post-stimulus) partial masking. The first is more pronounced in music and influences the transparency of the presentation. The second is related to the reverberation time. Dependencies of partially masked loudnesses are outlined quantitatively using abstract and musical sounds. A videotape presentation will illustrate these effects using the temporal variation of specific loudness-critical band rate distributions outlined on the screen of these loudness meters.

9:00

FFF2. The communication of dynamics between musicians and listeners through musical performance. Toshie Nakamura (Department of Psychology, Osaka University, Toyonaka, Osaka, Japan)

Musicians perform music according to their own interpretations. How does a player realize his/her interpretation of a piece of music in the form of tones? How is his/her intention understood by a listener? The relations among the performers' intentions, intensity of tones produced by the performers, and listeners' perception were investigated quantitatively. The noteworthy findings were as follows. (1) *Crescendo* was easier to recognize than *decrescendo*. (2) It appeared that rising pitch enhanced an impression of *crescendo*, and falling pitch enhanced that of *decrescendo*. (3) The influence of "impression of the end" on *crescendo* was suggested. These findings were confirmed in the experiments by means of synthesized sound sequences and under well-controlled conditions.

9:30

FFF3. Dynamics of musical expression. Edward C. Carterette and Roger A. Kendall (Departments of Psychology and Ethnomusicology and Systematic Musicology, UCLA, Los Angeles, CA 90024)

Various aspects of the dynamics of musical expression in terms of both acoustic and perceptual parameters were studied. In one study, duets of soprano orchestral winds performed brief musical passages in unison and in simple diatonic harmony. Perceptual spaces, obtained by scaling, were compared with acoustical analyses. Orderings of instrument pairs in three-dimensional space were influenced by context (unison versus harmonized single notes and melodic fragments) and by such acoustic factors as combinations of excitation mechanisms and tone-hole lattice cutoff frequencies. In a study of musical expression, single instruments, which differed in performer degrees of freedom (piano, violin, trumpet, clarinet, and oboe), were played at three levels of expressiveness. The aim was to explicate the variables used by the performer in successfully communicating expressive intent. Acoustical analyses were compared by means of a theoretical model based on the interactions of pitch, time, and dynamic contour strata.

10:00

FFF4. How to treat dynamic range if there is no dynamic space. Adrianus J. M. Houtsma (Institute for Perception Research IPO, P. O. Box 513, 5600 MB Eindhoven, The Netherlands)

The large dynamic range of modern sound equipment is often incompatible with the playback situation. Compact-disc music played at a low level because of sensitive neighbors, or car-radio music played against a background of intense and varying noise, is almost always partly inaudible unless signal levels are constantly adjusted. For a fixed ceiling on sound dynamics or for constant levels of background noise, as is often found in a typical home, use of dynamic compression according to the function $y = Ax^m$ turns out quite satisfactory, with the compression exponent m between 1 (no compression) and $2/3$ (maximum acceptable compression), and attack and release times equal to 5 and 180 ms, respectively. For treatment of music and speech under conditions of intense and variable noise, as is the case in a moving car, several signal processing schemes are discussed, such as noise-dependent volume control, noise-dependent frequency control, and noise-dependent amplitude compression.

10:30

FFF5. Musical dynamics with keyboard instruments. Donald E. Hall (Physics Department, California State University, 6000 J Street, Sacramento, CA 95819)

Keyboard instruments provide the extreme cases of wide or narrow dynamic ranges. The harpsichord mechanism apparently has a fixed dynamic level, aside from registration changes. But performers do claim to have some expressive control, and it may be asked whether measureable physical differences can be demonstrated. The pipe organ uses both swell box shades and registration changes, and what may be typical sound level differences involved will be discussed. The clavichord and piano are both more akin to other instruments in having output strongly dependent on the force with which the keys are struck. Comparisons among these cases raise questions about how much perceived musical dynamics depends on mere loudness, and how much on timbre, articulation, and other aspects of the musical performance.

FFF6. Distinguishable musical dynamics. Blake R. Patterson (Network Management Division, AT&T Bell Laboratories, 480 Red Hill Road, Middletown, NJ 07748)

Loudness changes, which composers specify by musical dynamics *pp*, *p*, *mp*, *mf*, *f*, and *ff*, make music exciting. The goal of this study is a usable and rational specification of an "acceptable" dynamic range for musicians. Requiring that a note at one dynamic level be at least s dB more intense than nearby notes at the next-lower dynamic level leads to a minimum solo-instrument per-note dynamic range $(k-1)(d+s)$, where k is the number of dynamic levels and d is the maximum intensity difference between nearby notes at "constant" dynamic level. For six dynamic levels and the bassoon, for which d is about 5 dB, a 1-dB spacing s between dynamic levels gives a minimum dynamic range of 30 dB ["Musical Dynamics," *Sci. Am.* 231 (5), 78-95 (November 1974)]. Some related results published by others over the past 14 years are briefly noted. Professional performances having different dynamic ranges are exhibited. A live demonstration of distinguishable dynamics on the bassoon is presented. Much remains to be studied in this rich field, for example, (1) dependence of d on instrument and dynamic level; (2) dependence of (listener-perceived) loudness on musical factors other than intensity, such as pitch, timbre, note duration, and transients; (3) dynamic complexities of ensemble performance: solo levels, heterogeneous ensemble, "chorus" effects, ensemble with soloist, etc.; (4) dependence of musical dynamics on music type, performance environment, and performance objective.

FFF7. Influence of hammer velocity in piano sound. Caroline Palmer (Psychology Department, Ohio State University, Columbus, OH 43210) and Judith Brown (Physics Department, Wellesley College, Wellesley, MA 02180 and MIT Media Lab E15-483, Cambridge, MA 02139)

Recent models of piano string excitation suggest that string motion is proportional to hammer velocity. Preliminary to measuring dynamics in piano performance, the relationship between radiated piano sound and hammer velocities was studied. A computer-monitored piano allowed reproduction of a range of hammer velocities, and a microphone placed near a pianist's head location recorded radiated sound. The first experiment indicated that peak amplitude of the waveform was directly proportional to the hammer velocity of individual notes for a wide range of frequencies and hammer velocities. A second experiment investigated the waveforms of notes struck simultaneously and compared these with the waveforms of the individual component notes. The peak amplitude of the dyads was directly proportional (with proportionality close to 1) to the sum of the individual peak amplitudes. A third experiment extended these findings to a wider range of dyadic pitch intervals. These findings suggest linear relationships between hammer velocity, amplitude of string motion, bridge motion, and amplitude of the soundboard vibration for single notes. The additivity of dyads indicates that the principle of superposition is obeyed with a small energy loss. [Work supported by MIT Media Lab, Ohio State University Small Research Grant, and a Wellesley College Brachman Hoffman Fellowship.]

FRIDAY MORNING, 26 MAY 1989

NEWHOUSE I, ROOM A1, 9:00 TO 11:15 A.M.

Session GGG. Psychological Acoustics IX: Complex Stimuli

Arlene E. Carney, Chairman

Boys Town National Institute, 555 North 30th Street, Omaha, Nebraska 68131

Contributed Papers

9:00

GGG1. Pitch of complex tones with many high-order harmonics. Adrianus J. M. Houtsma (Institute for Perception Research IPO, P. O. Box 513, 5600 MB Eindhoven, The Netherlands) and J. Smurzynski (Division of Otolaryngology/Surgery, University of Connecticut Health Center, Farmington, CT 06032)

Pitch identification and pitch discrimination experiments were performed for complex tones with missing fundamentals between 200 and 300 Hz and with many successive harmonics varying from low (below the

10th) to high (above the 25th) harmonic order. Identification performance was found to degrade with increasing harmonic order from an essentially perfect to an asymptotic level that was clearly less than perfect but much better than chance. Just-noticeable differences in (missing) fundamental frequency were found to increase, with increasing harmonic order, from a fraction of 1 Hz to an asymptotic level of about 5 Hz. Influence of phase was found only for tone complexes of high harmonic order. Results support the existence of two separate pitch mechanisms in the auditory system, one based on pattern matching of resolved frequencies, the other on periodicity of nonresolved frequencies.

GGG2. From acoustics to the social sciences: The concept of "sound effects." Jean-Paul Thibaud (201 East 19th Street, Apartment 9B, New York, NY 10003)

The analysis of sound within context is very complex. It requires an interdisciplinary approach in several fields: physical, architectural, psychological and physiological acoustics, and environmental sociology. The concept of "sound effect" involves concrete links between various forms of data. Sound effect can be defined by three major characteristics. (1) It is not the product of a cause, but rather the product of the relationship between the signal and the context (i.e., Doppler effect). Thus its definition necessitates a recognition of the physical signs in question and of the conditions of propagation and sound perception. (2) It involves a modal or instrumental logic. (3) It permits a general discourse about sound, but requires examples related to a particular situation. The concept of sound effect serves as a tool to uncover information about sound. It allows the researchers to put the sound phenomena in a wider perspective. [Work supported by Centre National de la Recherche Scientifique (CNRS, France).]

GGG3. Comparing monotonic, diotic, and dichotic presentation modes in synchrony detection. Virginia M. Richards (Psychoacoustics Laboratory, Department of Psychology, University of Florida, Gainesville, FL 32611)

In a 2IFC paradigm, listeners discriminated between sounds composed of two bands of noise whose center frequencies were 2500 and 5000 Hz, and whose envelopes were either identical or time delayed, one relative to the other. The extent of the temporal delay necessary for 70% correct discriminations was estimated. Dichotic masking noise was continuously present. Four presentation modes were employed: diotic, monotonic, dichotic, and monotonic with contralateral maskers. In the monotonic with contralateral masker condition, bands centered at 2500 and 5000 Hz were presented to the ear contralateral to the signal bands. Those "masker" bands were independently chosen, and so were unrelated to each other and the signal bands. Performance levels were best in the diotic condition, an average delay of 3 ms being needed for discrimination. Performance levels in the dichotic, monotonic, and monotonic with contralateral masker conditions were only slightly poorer. [Work supported by the Air Force Office of Scientific Research and a National Institutes of Health post-doctoral fellowship.]

GGG4. Signal parameters that reduce masking produced by multicomponent simultaneous maskers of uncertain frequency. Donna L. Neff (Boys Town National Institute, 555 North 30th Street, Omaha NE 68131)

Large amounts of simultaneous masking can be produced by changing the spectral content of multicomponent maskers with each presentation. This experiment examined whether simple changes in signal properties could reduce this masking. The number of masker components varied from 2 to 100 across conditions. Thresholds for simultaneous masking were measured for four listeners for a 1000-Hz sinusoidal signal equal in duration to the masker (200 ms) and compared to thresholds measured as signal duration was shortened and signal position varied. The largest release from masking occurred for the shortest (10-ms) signal temporally centered in the masker. Dichotic presentation also decreased thresholds relative to diotic or monaural conditions. In contrast, the use of different signal spectra, specifically a narrow-band noise, AM, or QFM signal, all centered at 1000 Hz, did not produce consistent reductions in masking. Overall, the results indicate that changes in the temporal or spatial relation of masker and signal are much more effective in reducing masking than changes in frequency composition under conditions of masker frequency uncertainty. [Work supported by AFOSR.]

GGG5. Forced-choice discrimination of complex tones. Annabel J. Cohen (Department of Psychology, Dalhousie University, Halifax, Nova Scotia B3H 4J7, Canada)

The study is part of a research program examining the effects of small integer frequency ratio relations on memory for unfamiliar (microtonal) sets of tones. Previous work indicated the benefit of both successive and simultaneous small integer context in an absolute judgment task. In order to determine whether such benefits were cognitive as opposed to sensory in origin, a quasifixated-standard two-alternative forced-choice discrimination task was conducted that had lower demands on memory than the absolute judgment task but could potentially lead to the same context effects. Discrimination of nine complex tones of a microtonal scale ranging from 545 to 636 Hz (tones separated and discriminanda differing by one-third semitone) was tested under simultaneous and successive context conditions. Preliminary data indicate that musically trained listeners, compared to untrained, showed a small but significant benefit from a dichotic simultaneous bass tone having the ratio 2:3 with one central tone of the nine-tone scale but not from a dichotic tone having a Unison relation with the central tone. Successive (melodic) context effects were less apparent in this discrimination study than in the absolute judgment experiments.

GGG6. Detection and recognition of temporal patterns. Daniel L. Weber (Department of Psychology, Wright State University, Dayton, OH 45435)

The detection/recognition theorem [S. J. Starr, C. E. Metz, L. B. Lusted, and D. J. Goodenough, *Radiology* **116**, 533-538 (1975)] was used to predict a listener's ability to recognize (identify) one of a pair of temporal patterns based upon the listener's detection performance. The six experimental conditions compared all possible pairs of four 860-ms patterns. The four patterns consisted of either two or four 1-kHz sinusoidal components of either 20- or 200-ms duration. Specifically, the four patterns were: two 20-ms components separated by 820 ms; two 200-ms components separated by 460 ms; four 20-ms components separated by 260 ms; and four 200-ms components separated by 20 ms. All components were square gated on and off at zero crossings. In four of these comparisons, stimuli differed on "one" dimension (either number or duration of components), whereas the other two comparisons involved differences in "two" dimensions (both number and duration of components). (Interstimulus gaps were necessarily confounded with number of components and component duration in all conditions.) Although the results were not uniform for all 11 listeners, comparisons that involve two dimensions generally led to better recognition performance than comparisons that involved one dimension. More surprising was the apparent unimportance of temporal structure for the encoding of these complex stimuli inferred from the general failure to reach predicted recognition performance. [Research supported by AFOSR through WPAFB AAMRL/BBA.]

GGG7. Effects of phase changes in low-numbered harmonics on formant frequency matches. J. Denis McKeown and Christopher J. Darwin (Laboratory of Experimental Psychology, University of Sussex, Brighton BN1 9QG, England)

Phase changes to single harmonics (of a 125-Hz fundamental) in the first formant (F_1) region can change the phoneme boundary between /l/ and /e/ along an F_1 continuum in an identification task [C. J. Darwin and R. B. Gardner, *J. Acoust. Soc. Am.* **79**, 838-845 (1986)]. Physiological correlates of this effect have been noted in the ALSR response of guinea-pig auditory nerve fibers [A. R. Palmer *et al.*, in *The Psychophysics of Speech Perception*, edited by M. E. H. Schouten (Nijhoff, Dordrecht, 1987)] and in an auditory model with synchrony suppression. The present work demonstrates similar phase effects using a matching paradigm.

Subjects matched sounds along a continuum that differed in $F1$ frequency and could have additional formants. The target sounds (fundamental = 125 Hz; $F1$ = 440 Hz) were taken from the matching continuum but had the phase of a single harmonic changed. Systematic and consistent differences in the matched $F1$ value were found as a function of phase. Although the starting phase of the components influenced the size of the matched difference in $F1$, the difference was still present for randomized starting phases. The effect required at least three frequency components and varied with fundamental frequency.

10:45

GGG8. Temporal window shape as a function of frequency and level. Christopher J. Plack and Brian C. J. Moore (Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge CB2 3EB, England)

The temporal window is an intensity weighting function that processes the internal representation of auditory stimuli by integrating energy over time. Temporal window shapes were measured using the technique described in an earlier article [Moore *et al.*, *J. Acoust. Soc. Am.* **83**, 1102–1116 (1988)]. Window shapes were measured at four different frequencies (300, 900, 2700, and 8100 Hz) and at three different masker levels covering a 20-dB range at each frequency. The shape of the temporal window was well described by modeling each side as the sum of two rounded-exponential functions. The equivalent rectangular duration (ERD) of the window was roughly constant at about 8 ms for center frequencies from 900 to 8100 Hz, but increased to about 13 ms at 300 Hz. The increase at 300 Hz may be explicable in terms of "ringing" in the auditory filter. The ERD decreased somewhat with increasing level, for

example, having a value of about 10 ms at 2700 Hz with a 20-dB masker spectrum level and about 7 ms with a 40-dB masker spectrum level. Temporal window shapes can be used to produce temporal excitation patterns that illustrate the effect of limited temporal resolution on the internal representation of acoustic stimuli. [Work supported by MRC, UK.]

11:00

GGG9. Perceiving fractal noises in pitch, loudness, and duration. Mark A. Schmuckler and David L. Gilden (Department of Psychology, University of Virginia, Charlottesville, VA 22903)

Three experiments examined discrimination among a family of fractals possessing different power law spectra. In experiment 1, random numbers sequences were generated so that their spectra had slopes (in log-log plane) of 0 (white noise), -1 (flicker noise), and -2 (brown noise). The distributions were used to produce sequences that could vary in pitch, loudness, or duration. Listeners categorized each sequence as being of one of the three noise types. Thurstonian scaling revealed that noise segment classification could be accomplished on the basis of pitch and loudness variation, but not on the basis of duration changes. Experiment 2 explored simultaneous pitch and loudness variation, finding dimensional redundancy for white/flicker discrimination suggestive of a pitch-loudness interaction. Experiment 3 examined sensitivity for discrimination of noises as a function of the power law exponent. Listeners heard two sequences of tones and indicated whether they had the same or different power law exponent. The range of exponents spanned the interval $[0, -3]$, with the stimuli under comparison differing by 0.2, 0.4, 0.6, or 0.8. ROC analyses revealed some variation in sensitivity to the fractal exponents as a function of position in the $[0, -3]$ interval.

FRIDAY MORNING, 26 MAY 1989

REGENCY C, 9:00 A.M. TO 12:00 NOON

Session HHH. Underwater Acoustics VIII: Sea Surface Noise I

Herman Medwin, Chairman

Department of Physics, Naval Postgraduate School, Monterey, California 93943

Chairman's Introduction—9:00

Invited Papers

9:05

HHH1. Noise generation by newly created bubbles. Michael S. Longuet-Higgins¹ (La Jolla Institute, 7855 Fay Avenue, La Jolla, CA 92037)

Air bubbles trapped at a free surface by breaking waves, or emerging from an underwater nozzle, or splitting, or coalescing in various ways, all undergo strong "shape oscillations" before settling into a spherical shape. In a fully nonlinear theory (not the usual linearized approximation), the shape oscillations can be shown to emit a strong monopole radiation of sound. Second-order calculations suggest that this is a significant, perhaps dominant, source of bubble noise in the ocean. Resonance between the second harmonic of each shape oscillation and the radial "breathing" mode of a given bubble leads to enhanced emission of sound at or near the breathing-mode frequency. There can also be coupling between a bubble and the free surface from which it is detached. Oceanic data and laboratory experiments relating to these predictions will be discussed, particularly the well-known experiments by Fitzpatrick and Strasberg (1957); also, broadband spectra of oceanic noise from a variety of locations will be discussed. ¹ On leave from Department of Applied Mathematics and Theoretical Physics, University of Cambridge, England.

9:25

HHH2. Ambient sound beneath breaking surface waves. David M. Farmer (Institute of Ocean Sciences, P. O. Box 6000, Sidney, British Columbia V8L 4B2, Canada) and Svein Vagle (Department of Physics, University of Victoria, P. O. Box 1700, Victoria, British Columbia V8W 2Y2, Canada)

Ambient sound measurement obtained beneath breaking surface waves over the frequency range 44–20 000 Hz are described. The observations are obtained from an instrument located 24–32 m beneath the surface during experiments both in open ocean and continental shelf environments. Individual breaking events are positively identified using underwater video photography, and it is confirmed that they radiate sound throughout the observed frequency range. A distinctive feature of the acoustic signature is the presence of spectral peaks, the frequency of which may remain generally consistent from one breaking event to the next, but which can change significantly over the course of a storm or from one storm to another. A theory is proposed based on the concept of trapping of a portion of the sound in the surface waveguide formed by the ocean surface bubble layer. Simultaneous measurements of the bubble population and size distribution obtained with a multifrequency inverted echo sounder allows calculation of the resulting (dispersive) sound-speed anomaly profile. Comparison of the observed spectral peak frequencies with those predicted yields good agreement.

9:45

HHH3. Bubble sources of the Knudsen spectra of ocean noise. Herman Medwin (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

The source levels, frequencies, and damping constants of individual bubbles, catastrophically generated by spilling breakers, have been measured in the NPS Ocean Acoustic Wave Facility, OAWF. OAWF consists of a 17-m-long \times 1-m-deep \times 1-m-wide water channel in which waves are driven by an oscillating plunger. The waves grow and then form spilling breakers on the surface of a large, nonreverberant volume at the end of the channel. Two calibrated hydrophones suffice to identify the type of bubble, its surface location, and its dipole axial source strength. The sources of the breaker noise have thereby been identified as transient bubbles of resonance frequency 350 Hz to 50 kHz, with lifetimes from 2 to 20 ms. These bubbles radiate as dipoles from positions within a few hundred μ m to a few mm of the surface. The long-term average of the bubble radiations has the same slope as the Knudsen wind wave spectra at sea, 5 dB/oct. [Work supported by the Office of Naval Research.]

10:05

HHH4. Underwater sound from whitecaps at sea. Reginald D. Hollett (SACLANT Undersea Research Centre, I19026 La Spezia, Italy)

Observations are presented of the underwater sound from whitecaps at sea. A vertical array of hydrophones was used to form an end-fire beam towards the sea surface, receiving the sound from a patch of the surface above the array. At the same time, the surface patch was visually monitored using a video-recording facility. The time series of the sound from the patch was spectrally analyzed at intervals of a fraction of a second, over the band from 187.5 to 1500 Hz. The resulting time variation of the sound spectrum shows distinct bursts of sound. The strong bursts coincide with visual detections of whitecaps above the array. The burst of sound from a whitecap lasts for several seconds and is seen to be associated with the breaking phase. The sound spectrum during the breaking phase has notable low-frequency content, the spectrum appearing almost flat from 187.5 Hz up to about 1000 Hz.

10:25

HHH5. Some statistics of sea noise. P. A. Crowther (Marconi Underwater Systems Limited, Croxley Mill, Watford WD18YR, United Kingdom)

Surface-derived ambient noise has been measured on a long-term basis by the use of a bottom-mounted narrow beam sonar, in depths of 65–80 m, over windspeeds 3–20 m/s, in frequency bands over 6–50 kHz. Results are presented as time histories and derived statistics. It is found that, after detection and smoothing, the noise power in a narrow beam fluctuates widely, "spikes" in time being identifiable with individual breaking waves. The statistics of fluctuation is found to depend on beamwidth, the coefficient of variation being typically 10/beamwidth in meters. Time autocorrelation functions of noise power are presented, which also depend on beamwidth and windspeed. The mean horizontal speed of the noise sources can be deduced and is approximately half the windspeed in a well-developed sea. These results are interpreted in the light of known spectra of sea waves, and simple assumptions about the nature of the wave breaking process.

Contributed Paper

10:45

HHH6. *In situ* acoustic signature of low sea state microbreaking. Garr E. Updegraff and Victor O. Anderson (University of California, San Diego, Marine Physical Laboratory of the Scripps Institution of Oceanography, San Diego, CA 92152)

At sea acoustic and video observations of low sea state microbreaking were carried out with a subsurface instrument in the summer of 1988. Correlation of subsurface video images of ripples and their acoustic signals will be presented. A four-hydrophone array provides spatial localizations of individual bubble signatures. Spectral and intensity distributions of the bubble signatures will be presented.

11:00–12:00

Panel Discussion

Chaired by: David Farmer
Institute of Ocean Sciences, Sidney, BC, Canada

Session III. Speech Communication XI: Production, Laryngeal Effects, Kinematics, and Intonation

Lawrence Rosenblum, Chairman

Department of Psychology, Wellesley College, Wellesley, Massachusetts 02181

Contributed Papers

1:15

III1. Perspectives on sound in Sanskrit literature on natural philosophy. M. G. Prasad (Department of Mechanical Engineering, Stevens Institute of Technology, Hoboken, NJ 07030)

It is known that the literature in Sanskrit language that includes "Vedas and Upanishads" represents some of the earliest known references dealing with various topics of natural philosophy and science. The historical background of these references has been estimated to be of several thousand years BC. In spite of the lack of a precise historical account of the development of the literature in Sanskrit language, there have been many works of translation from Sanskrit to several European languages including English. The topic of "sound" seems to have received fundamental importance in Sanskrit literature in the areas of both natural philosophy, science, and art. This paper describes the perspectives on sound as found in Sanskrit literature particularly in "Vedas and Upanishads." The paper also discusses some of the important applications of the knowledge and perspectives on sound that are reflected in acoustic phonetics of Sanskrit language, Vedic chants, and classical music and dance of India.

1:27

III2. Acoustical manifestations of velopharyngeal closure during swallowing. Sandra L. Hamlet, Duane M. Smith (Department of Otolaryngology, Wayne State University, Detroit, MI 48201), and Lewis Jones (Department of Radiology, Harper Hospital, Detroit, MI 48201)

In normal swallowing, velopharyngeal (VP) closure transiently isolates the nasopharynx from the aerodigestive tract for a period of approximately 700 ms. Although there are intrinsic sounds generated during swallowing that can be recorded from the throat surface [Hamlet *et al.*, J. Acoust. Soc. Am. Suppl. 1 83, S23 (1988)], VP closure is ordinarily a silent event requiring special techniques for its acoustical detection. A 500- or 1000-Hz tone is introduced into one nostril, and its presence is detected using a miniature accelerometer taped to the throat just below the angle of the mandible. During VP closure, there is attenuation of this component of the accelerometer signal recorded during swallowing. The onset of the period of attenuation corresponds to the onset of velar closure determined radiologically. The time of VP reopening can be detected with less certainty, possibly owing to resonance effects, and the position of the epiglottis and base of the tongue at that moment in some subjects. Medical applications of this procedure are discussed. [Work supported by NIH.]

1:39

III3. Pressure-flow relationships in the larynx. Marshall E. Smith and Gerald S. Berke (Division of Head and Neck Surgery, 62-132 CHS, UCLA Medical Center, 10833 Le Conte Avenue, Los Angeles, CA 90024-1624)

Measurements of air pressure and flow were made using an *in vivo* canine model of the larynx. Subglottic and supraglottic pressures at vary-

ing flow rates were taken during phonation, induced by laryngeal nerve stimulation. Results shows that, during constant vocal fold stiffness, subglottic pressure did not rise significantly with increased airflow. Increasing flow rate was associated with an increase in the open quotient as measured glottographically. Data from this experiment were compared with the theoretical two-mass model of the larynx. Model parameters were adjusted to approximate the canine vocal folds and the *in vivo* experimental pressure-flow relationships were simulated. The larynx in the *in vivo* canine model exhibits flow-dependent decreasing impedance during phonation in that increasing flow rate is not accompanied by increases in subglottal and transglottal pressure. [Work supported by VA Technical Merit Review Grant.]

1:51

III4. An experimental design for testing the validity of the collapsible tube model of the larynx. Gerald S. Berke, David P. Arnstein, Marshall E. Smith, Manuel Natividad (Division of Head & Neck Surgery, 62-132 CHS, UCLA Medical Center, 10833 Le Conte Avenue, Los Angeles, CA 90024-1624), and William A. Conrad (30 West 71st Street #3D, New York, NY 10023)

The *in vivo* canine model of the larynx was used to measure transglottic pressures and airflow during phonation. Direct intraglottal pressures were measured at various positions within the glottis. Conditions of supraglottal resistance were also simulated. Pressure drop-flow curves were compared with data on collapsible tubes. The *in vivo* canine model of the larynx demonstrates a number of the described features similar to oscillation in collapsible tubes. [Work supported by VA Technical Merit Review Grant.]

2:03

III5. /s/ and /j/ as a function of linguapalatal groove place and width. S. G. Fletcher (Department of Biocommunication, UAB, University Station, Birmingham, AL 35294) and D. G. Newman (University of Queensland, 6 Rock Street, St. Lucia 4067, Queensland, Australia)

Palatometric feedback was used to guide three normal talkers in systematically varying the place and width of the sibilant groove. Linguapalatal contact was manipulated to produce 2- to 12-mm-wide grooves in each of seven-eight rows of contact sensors spanning the front third of the palate. Both the rows and the sensors in each row were 2 mm apart. The subject's task was to establish a stipulated groove, then channel air through it to produce a "hissing" noise. These responses were tape recorded then later low-pass filtered, digitized, and shaped to fit normal sibilant intensity, duration, and onset/offset times. The resultant sounds were played ten times each in random order to 14 listener judges in individual listening sessions. The listeners were instructed to label each sound heard as "definitely an 's,'" "probably an 's,'" "probably an 'sh,'" or "definitely an 'sh.'" Statistical analysis of these observations yielded main effects for both groove width and place. The findings will be discussed from the viewpoints of biomechanical and auditory shaping hypotheses. [Work supported by NIH Grant NS 24697.]

III6. Alterations in lower lip and jaw kinematics of stutterers and nonstutterers as a function of speech rate. Luc F. De Nil and James H. Abbs (University of Wisconsin, Speech and Motor Control Laboratories, Madison, WI 53705)

Previous articulatory kinematic studies of stutterers and normally fluent speakers during perceptually fluent speech have revealed subtle, but possibly significant, between-group differences. Such observations lend support to the hypothesis that stutterers and nonstutterers somehow differ in their basic motor control ability for speech. Comparing the influence of speech rate on stutterers' and nonstutterers' articulatory kinematics would provide a further test for this hypothesis. To this end, seven adult stutterers and seven normal speakers produced repetition sequences of the syllable /bae/ at six different rates. Multiple kinematic measures of the lower lip and jaw movements during oral closure were applied. Peak velocity and displacement were observed to differ to subtle ways between the two groups, with greater differences apparent at faster speaking rates. Stutterers also differed from normal speakers in timing of peak velocities at various speech rates. The results provide additional information on the possible role of motor control processes in stuttering. [Work supported by NIH Grants NS-13274 and NS-16373.]

III7. Electropalatographic evidence for long-range coarticulatory effects in VCVCV sequences. Daniel Recasens (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

This study reports some electropalatographic data on tongue dorsum contact showing coarticulation to extend up to three or four phonetic segments away from the target gesture. Coarticulatory effects in degree of palatal contact for V = [i] vs [a] and for C = [f] vs [t] were analyzed along all possible VC[ə]CV combinations of those same vowels and consonants. It was predicted that the temporal domain of coarticulation would vary according to the degree of tongue dorsum contact involved during the production of the contextual phonemes. The findings show that, in general, the coarticulatory effects are inversely dependent on the degree of tongue dorsum contact for those phonemes that are immediately adjacent to the target (e.g., C2 for the V3-dependent anticipatory effects). Moreover, carryover effects reached the segments located at the other side of [ə] in specific circumstances (e.g., when C2 = [t]). The theoretical implications of the results of this experiment are discussed in light of recent findings in the literature.

III8. Phonetic product estimates of infant phonetic development. Harold R. Bauer (Division of Speech and Hearing Science, The Ohio State University, Columbus, OH 43210) and Michael P. Robb (John A. Burns School of Medicine, Division of Speech Pathology and Audiology, University of Hawaii, Honolulu, HI 96822)

The phonetic development of six children was estimated across the first 2 years of life. Using a phonetic product measurement scheme

(Bauer, 1988), the children's monthly vocalization samples were analyzed for phonetic contrast. The number of observed phones in five different consonantlike (place of articulation) and three vowellike (tongue advancement) categories was multiplied to yield a phonetic product for each vocalization. The results were summarized and plotted for the six children's monthly vocalization samples with microcomputer software. The results indicated a general increase in the phonetic product of vocalizations as a function of chronological age. Individual trends for each child are discussed in relationship to the general function in our search for general measures spanning the prelinguistic-to-linguistic period.

III9. On /r/-/w/ substitution in child speech. Kevin G. Lindland and Elzbieta B. Slawinski (Department of Psychology, The University of Calgary, Calgary, Alberta T2N 1N4, Canada)

The role of phonetic context in /r/-/w/ substitution errors was investigated. The subject population consisted of 20 children with normally developing articulation skills, 33 to 52 months in age. The subjects were given a minimal pairs discrimination task in order to assess the accuracy of their perception for "r" and "w" consonants. Subsequently, the subjects were administered a test of production in which they had to identify and produce the names of objects drawn from a common set presented to them. Acoustical analysis of the misarticulated /r/ reveals three patterns, which can be distinguished acoustically by different formant values and/or their changes in time. Results indicate that substitution errors are highly context dependent, with the percentage errors for the entire population ranging from 0% when /r/ was in the final position to 36% in the initial position in the word. These findings suggest the possible importance of a cognitive strategy, as well as the presence of shifted categorical boundaries. Results are discussed in terms of the developing child's limited mental resources and perceptual skills.

III10. Influence of national languages on learning English. Pan Jianping (Department of Information Technology in Education, East China Normal University, North Zhongshan Road, Shanghai 200062, People's Republic of China)

Every language can be described in terms of a set of phonemes, including, in part, vowels, diphthongs, semivowels, and consonants. In China, Mandarin is the standard spoken Chinese, i.e., the national language. There are many additional dialects, each with its own set of phonemes. Fluent speakers in Mandarin and another dialect can therefore pronounce two sets of phonemes. The phonemes are produced mainly by operations of the vocal tract. The connections between vocal tract movements and phonemes are established in childhood, first for the dialect and then for Mandarin. This led us to compare the ability to learn to speak English among Chinese speakers using different dialects and therefore different sets of phonemes. It was observed that the greater the similarity of the dialect to English, the more intelligible and natural were the initial attempts to speak English. This was in spite of similar fluency in the national language, Mandarin. It was concluded that early learning of English was important, independent of the national language, and it was hypothesized that similar results would be obtained in other non-English-speaking countries.

III11. Glottal source estimation using a sum of exponentials model. A. K. Krishnamurthy (Department of Electrical Engineering, The Ohio State University, 205 Drees Laboratories, 2015 Neil Avenue, Columbus, OH 43210)

This paper describes an algorithm for simultaneously estimating the parameters of a model for the glottal source and the vocal tract filter. The glottal source signal is described by the four-parameter LF model [Fant *et al.*, Speech Transmis. Lab. Q. Prog. Stat. Rep. (1984)]. The vocal tract is modeled as an all-pole filter; however, to allow for the effects of source-tract interaction, a separate model is used in the closed and open glottal phases. Under these assumptions, it is shown that the output speech signal can be modeled as a sum of complex exponential signals. The parameters of the speech signal model are estimated using the algorithms described in Kumaresan and Tufts [IEEE Trans. Acoust. Speech Signal Process. ASSP-30, 833-840 (1982)]. The electroglottograph signal is used to obtain a preliminary estimate of the location of the closed and open phases in each pitch period. The analysis of steady vowel sounds from several speakers indicates that this method provides a very accurate estimate of the glottal source and the vocal tract formants. [Work supported by the OSU Seed Grant program.]

3:42

III12. The effect of macroprosodic context on consonant perturbations of fundamental frequency. John Kingston (DMLL, Morrill Hall, Cornell University, Ithaca, NY 14853-4701)

Kohler argues, on the basis of German data, that differences in *F0* before or after voiced and voiceless stops (the *microprosodies*) may be obliterated by the larger scale effects on *F0* of the surrounding intonation contour (the *macroprosodies*) [Kohler, *Phonetica* 39, 199-218 (1982)]. In a similar fashion, Silverman claims, on the basis of English data, that the direction of *F0* change after stops differing in voicing depends on the surrounding macroprosodic contour [Silverman, *Phonetica* 43, 76-91 (1986)]. The macroprosodic contour in which word-initial consonants [p, t, b, d, sp, n] were embedded at two levels was manipulated: (1) The consonant occurred either immediately before a stressed vowel or before an unstressed vowel, e.g., *palace* versus *police*; and (2) the words occurred either in focus or with focus on the preceding word, e.g., *I saw a few churches on Monday, but many PALACES on Wednesday* versus *I saw only a few on Monday but MANY palaces on Wednesday*. These manipulations were entirely orthogonal and allow us to examine both the effects of word internal as well as syntactically determined macroprosodies on the extent and direction of consonantal effects on *F0* in English.

3:54

III13. Voicing epoch determination with dynamic programming. David Talkin (AT&T Bell Laboratories, Room 2D-448, Murray Hill, NJ 07974)

During voiced speech, the point of maximum flow change in each glottal cycle corresponds to the point of maximum excitation of the vocal tract. Accurate, reliable detection of this "epoch" beginning (or end) is useful for pitch synchronous analysis/synthesis in a variety of contexts. Dynamic programming has been applied to correlation function peak selection [Secrest and Doddington, ICASSP-83] and lagged waveform

matching [Ney, IEEE Trans. SMC-12 (1982)] for *F0* determination with excellent results, but these techniques do not yield the epoch locations. The method described in this paper applies dynamic programming to select waveform maxima directly from a short-time-energy-normalized LPC residual. The cumulative path costs are normalized by the path length. Local costs are based on peak amplitude and quality, transition costs on period and pulse similarity. The output is the set of pulse locations that globally satisfy the cost constraints over all voiced regions. Data to be presented indicate that these estimates reliably match those obtained semiautomatically from simultaneously recorded electroglottograph signals.

4:06

III14. A new approach for speech dynamics studies. Mohamed Mrayati (Centre d'Etudes et de Recherches Scientifiques, BP 4470, Damascus, Syria), René Carré, and Eric Castelli (Institut de la Communication Parlée, 46 avenue Félix Viallet, 38031 Grenoble, France)

A new dynamic model for vocal tract configuration based on a theoretical quantification of dynamic area function $A(x,t)$ into distinctive regions having areas $S(r,t)$ is proposed. Articulation is transcoded into changes of these $S(r,t)$ through three modes [Mrayati *et al.*, Speech Comm. 7, 257-286 (1988)]. A dynamic control strategy of $S(r,t)$ is extremely simple to apply and correlates well with human speech organ musculature movements. This strategy transprojects the classical longitudinal displacement of the constriction into a transversal area function variation of $S(r,t)$. This dynamic strategy can incorporate easily context-dependent allophonic variations, such as: coarticulation, reduction, assimilation.... The distinctive regions, along the vocal tract, are associated with specific consonants and vowels. These new "phonetic regions" concept can be used to vectorially code target configurations for consonants and vowels. The new concept relates, in a simple fashion, changes in the acoustic domain to changes in regional cross-sectional areas. Some results on VV, CV, VC, and VCV command strategies are presented.

4:18

III15. Closant curve characteristics of infant vocalizations. Michael P. Robb (John A. Burns School of Medicine, Division of Speech Pathology and Audiology, University of Hawaii, Honolulu, HI 96822) and Harold R. Bauer (Division of Speech and Hearing Sciences, The Ohio State University, Columbus, OH 43210)

The "closant curve" reflects the change in ratio of the frequency of consonants (closants) to vowels (vocants) in early vocal and speech development of young children (Bauer, 1988). Closant curves were plotted for seven children's monthly vocalizations across the first 2 years of life. The closant curves were then compared to monthly data plots of each child's vocal fundamental frequency and vocalization duration. Assessment of developmental trends for the phonetic and acoustic data was based on visual examination of the data plots and curve fitting procedures. The resulting developmental functions are discussed in relation to the reported closant curves underlying environmental and physiological characteristics of language acquisition. These analyses provide an acoustic and phonetic framework from which to study early speech development. [Work supported by U. H. Research Council.]

Session JJJ. Physical Acoustics X: Inverse Problems in Acoustics II

Yves H. Berthelot, Chairman

*School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, Georgia 30332**Contributed Papers*

1:30

JJJ1. Determination of the elastic constants of anisotropic composite materials via laser photoacoustics. Bernard Castagnede, Wolfgang Sachse (Department of Theoretical and Applied Mechanics, Cornell University, Ithaca, NY 14853), and Michael O. Thompson (Department of Materials Science and Engineering, Cornell University, Ithaca, NY 14853)

This paper presents the solution of the inverse anisotropic medium problem in which the elastic constants of an anisotropic composite material are determined from ultrasonic wave speed measurements made in nonprincipal directions of a specimen. The ultrasonic waves were generated via the *point-source/point-receiver* technique using a 2-ns pulsed Nd:Yag laser as a source and a 2.5-mm-diam capacitive or a 1.3-mm piezoelectric transducer as a receiver. Data were acquired during an isoangular scan of the source relative to one of the principal acoustic axes of symmetry. In each waveform, the arrivals of the quasilongitudinal and the two quasishear bulk modes were measured. The elastic constants of the material were then recovered using an optimization algorithm. Experimental results are presented for a transversely isotropic composite material. It was found that the nonlinear fit between the experimental and the recovered slowness values is excellent. Some discrepancies are observed for the two shear modes. These are shown to be related to the complexity of the detected signals.

1:45

JJJ2. On the acoustic determination of the elastic moduli of anisotropic solids and acoustic conditions for the existence of symmetry planes. Andrew N. Norris (Department of Mechanics and Materials Science, Rutgers University, Piscataway, NJ 08855-0909)

The 21 elastic moduli of a homogeneous anisotropic solid can be determined from the second-order acoustical tensors associated with wave motion in six phase directions. The acoustical tensor can be calculated from the three phase speeds and displacement polarizations for a given phase direction. The six directions may be quite arbitrary as long as they cannot be contained in less than three distinct planes through the origin and do not all lie on the curves formed by the intersection of the unit sphere with an elliptical cone. It is shown, for example, how the general procedure can be used to find the elastic moduli for a thick plate specimen, for which acoustic access is limited to inspection from the two parallel faces. Acoustic conditions that are both necessary and sufficient, that a solid possess a plane of material symmetry, will also be discussed. [Work supported by NSF.]

2:00

JJJ3. GTD synthesis of resonance amplitudes in the backscattering from an elastic spherical shell. Steven G. Kargl and Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164-2814)

An elastic generalization of the geometrical theory of diffraction [P. L. Marston, *J. Acoust. Soc. Am.* **83**, 25-37 (1988)] was used to describe the backscattering of short tone bursts from an elastic spherical shell in water [S. G. Kargl and P. L. Marston, *J. Acoust. Soc. Am.* **85**, 1014-1028 (1989)]. In the present research, steady-state backscattering amplitudes are synthesized. The GTD model contains explicit terms for the specular reflection and individual Lamb surface wave contributions in a Fabry-Perot form. Numerical calculations compare the exact partial-wave series result with the GTD synthesis for the frequency range $7 < ka < 100$. These computations correspond to a stainless steel shell with an inner-to-outer radius ratio $b/a = 0.838$. An important feature of the GTD model is that only two parameters are needed to fully describe the individual Lamb wave resonance amplitudes. These parameters are the ratio of the Lamb wave phase velocity to the velocity of sound in water (c_l/c) and the radiation damping β_l . These should depend on local properties of the shell-water system. This simple parametrization of the direct scattering problem should facilitate novel approaches to the inverse problem. Identification of a scatterer may be achieved through an examination of the Lamb wave resonances via a numerical fitting of these parameters. [Work supported by ONR.]

2:15

JJJ4. Transverse Pearcey patterns observed in the reflection of ultrasound from curved surfaces. Carl K. Frederickson and Philip L. Marston (Department of Physics, Washington University, Pullman, WA 99164-2814)

This research concerns the reflection of light and sound from smooth curved surfaces to produce transverse cusp caustics (and associated wave fields) and the inference of shape information from such caustics and wave fields. From the generic wave front shape associated with transverse cusps [P. L. Marston, *J. Acoust. Soc. Am.* **81**, 226-232 (1987)], it has been calculated that reflection of sound from a point source off of a surface whose height $h(x,y) = h_1x^2 + h_2y^2x + h_3y^2$ (with $h_2 \neq 0$) produces a transverse Pearcey pattern [P. L. Marston, in *Acoustical Imaging* (Plenum, New York, 1988), Vol. 16, pp. 579-588]. The imaging of such patterns by raster scanning of a small hydrophone in water is described. The ultrasonic burst incident on the surface is sufficiently long to simulate the reflection of a steady-state sine wave. When the sound source was replaced by a light source (and the hydrophone replaced by a photodetector), the corresponding optical cusp caustic was imaged and used to infer the shape parameter h_2 for the polished metallic reflecting surface. These parameters facilitated a comparison of observed and predicted acoustic wave fields. [Work supported by ONR.]

2:30

JJJ5. Examination of finite amplitude ultrasonic pulses using light diffraction. J. W. Wolf (Department of Physics Ultrasonic Research Laboratory, Georgetown University, Washington, DC 20057)

Finite amplitude pulses are examined acousto-optically with a light diffraction apparatus. Using an empirical model of transducer frequency response, ultrasonic pulse Fourier spectra are derived for input to a light diffraction model. To then simulate harmonic distortion, the pulse spectra are used for input to a computational model based on the Burgers' equation for propagation of finite amplitude acoustic waves in a nonlinear medium. Several examples are presented that illustrate good agreement between experimental light diffraction patterns and those predicted by the pulse spectrum, propagation, and light diffraction theories. [Work supported by ONR.]

2:45

JJJ6. An equation for acoustic propagation in an inhomogeneous medium with relaxation loss. Adrian I. Nachman (Department of Mathematics, University of Rochester, Rochester, NY 14627), James F. Smith, III, and Robert C. Waag (Departments of Electrical Engineering and Radiology, University of Rochester, Rochester, NY 14627)

An equation for acoustic propagation in an inhomogeneous medium with relaxation loss is systematically derived from the classical dynamic equations together with an equation of state for relaxation. The derivation assumes small acoustic perturbations but accommodates arbitrary spatial inhomogeneities in material compressibility, density, and parameters of relaxation. The linearized wave equation obtained for n relaxation mechanisms has order $n + 2$, is causal, and yields the expected dependence of attenuation on frequency. Exact analytic expressions valid at all frequencies are given for the spatially varying attenuation coefficient, as well as phase velocity. A Green's function is calculated for the equation. The results may be used to model scattering for image reconstruction and the determination of statistical properties, such as average differential scattering cross section.

3:00

JJJ7. Extraction of information from low-frequency sounds generated within the human body. T. Douglas Mast and Allan D. Pierce (Graduate Program in Acoustics, Pennsylvania State University, 157 Hammond Building, State College, PA 16804)

The analysis of sounds generated within the body has been a staple of medical diagnostics since Laennec's invention of the stethoscope (c. 1804). Also, interest in muscle sounds dates back to Grimaldi's treatise on optics published in 1665. More recent work by Oster and Jaffe (1980), Barry (1987), and Frangioni *et al.* (1987) has led to quantitative insight on the physical processes that generate such sound. A feasibility analysis is presented here for the problem of source characterization using information taken outside the human body, with emphasis on sources of frequencies less than 250 Hz. The available information is the displacement time history of all points on the body's surface, although one would want to measure displacements at a restricted number of points. The body is modeled as a dissipative fluid-filled sphere or a finite cylinder. It is argued that sources of interest can be idealized as compact, rather than distributed, combinations of monopoles, dipoles, and quadrupoles. Thus the unknowns to be determined in regard to any frequency component are the source coordinates and the strengths and relative phasing of the probable multipoles. Results reported include measurement accuracies necessary for satisfactory source characterization. [Work supported by the William E. Leonhard endowment to Pennsylvania State University.]

3:15

JJJ8. Mixture composition determination from measurements of the acoustic nonlinearity parameter. E. C. Everbach and R. E. Apfel (Department of Mechanical Engineering, Yale University, P. O. Box 2159, New Haven, CT 06520)

An important inverse problem in biomedical acoustics is the determination of the composition of a tissue based upon measurements of its acoustic properties. Under many circumstances, the tissue can be modeled as an ideal mixture of components, each with known or assumed properties. The measured properties of the mixture as a whole are then related to those of the components through a set of mixture laws. Once formulated, the problem can be inverted to yield the volume or mass fractions of the components present in the tissue. In this laboratory, the use of the acoustic nonlinearity parameter B/A to help infer the composition of mixtures of biological materials has been studied. Various mixture models have been investigated, and a modified version of Apfel's model [J. Acoust. Soc. Am. 79, 148-152 (1986)] has yielded the most useful results. This modified mixture model will be presented, along with precise data that have been obtained on biological mixtures using a novel acoustic interferometry technique reported earlier [J. Acoust. Soc. Am. Suppl. 1 80, S4 (1986)]. [Work supported by the U. S. National Institutes of Health through Grant R04GM30419].

3:30

JJJ9. Biological cell characterization using inverse acoustic scattering. X. Chen and R. E. Apfel (Department of Mechanical Engineering, Yale University, P. O. Box 2159, New Haven, CT 06520)

The acoustic scattering properties of biological cells, especially red blood cells, have been studied both theoretically and experimentally. An apparatus consisting of two confocally positioned transmitter/receivers and a Coulter-type volume sensing zone was reported previously [J. Acoust. Soc. Am. Suppl. 1 84, S163 (1988)]. An inverse scattering theory incorporating the experimental data yields three bits of information for each cell: volume, density, and compressibility. Since both the density and compressibility of biological cells are almost linearly related to protein concentration, it might be possible to determine the protein concentration of these cells. The results obtained from several types of biological cells using this method will be presented. The changes in density and compressibility, and hence protein concentration, when cells undergo volume changes through swelling, shrinking, or heating will be assessed. [Work supported by the U. S. National Institutes of Health through Grant 5R01GM30419.]

3:45

JJJ10. Higher-order time-domain ultrasonic scattering. Thomas J. Cavicchi (Department of Electrical Engineering, University of Akron, Akron, OH 44325)

To improve the reliability of diagnosis of tissue pathologies (e.g., cancers) based on ultrasonic imaging, quantitative images of physiological parameters, such as speed of sound and/or absorption coefficient, can, in theory, be obtained by inverse scattering procedures. Previous studies [T. J. Cavicchi, S. A. Johnson, and W. D. O'Brien, Jr., IEEE Trans. UFFC-35, 22-33 (1988)] employed a moment method matrix formulation under a monofrequency assumption. Periodicities of the field phasors caused a nonuniqueness problem in the inverse scattering solution for scatterers with phase shift magnitudes greater than π ; in practical medical imaging, the phase shift would be hundreds of π . A reformulation of the method in the time domain appears more promising because the phase shifts are now merely time delays. The time-domain moment method formulation is presented for forward scattering. A preliminary numerical study has been undertaken using the moment method equations and the internal field calculated by the inverse discrete Fourier transform of the well-known monofrequency solution solved at many frequencies. Using the exact field within the integral produces a resulting good approximation of the exact field. Current studies are aimed at inversion for the forward-scattered field; future work will address inverse scattering.

Session KKK. Underwater Acoustics IX: Sea Surface Noise II

Herman Medwin, Chairman

Department of Physics, Naval Postgraduate School, Monterey, California 93943

Chairman's Introduction—1:30

Invited Papers

1:35

KKK1. Sound generated by fluid interfaces—Insights into sea surface noise. Douglas H. Cato (DSTO Maritime Systems Division, P. O. Box 706, Darlinghurst, NSW 2010, Australia)

A theory of sound generation by motion of fluid interfaces has been developed and applied to the sea surface. The only assumptions are that the density discontinuity at the sea surface can be modeled as a Heaviside function, and that the relationship $\partial p / \partial \rho = c^2$ applies within a fluid, where p is pressure, ρ is density, and c is sound speed. The theory shows that the motion of an interface generates sound with source strengths depending on the difference between the product of the density and the square of the sound speed on either side of the interface. Distributed monopole and dipole sources are possible. The insight this provides into sound generated by surface motion, bubble cloud oscillation, bubbles, droplets, and spray will be discussed. Most sources would be expected to show dipole directionality with axes close to, but not necessarily, vertical. Wind-induced turbulent pressure fluctuations across the sea surface are shown to be not a significant source of noise in the ocean, even though they have been considered as such.

1:55

KKK2. Low-frequency noise: Wave turbulence interaction and bubble cloud oscillations. W. M. Carey, D. G. Browning (Naval Underwater Systems Center, New London, CT 06320), and J. W. Fitzgerald (Kildare Corporation, New London, CT 06320)

Low-frequency ocean ambient noise data, when not dominated by shipping noise, show evidence for wind-dependent noise at frequencies less than 500 Hz. Vertical directionality measurements have a horizontal component with a broad frequency characteristic. This effect is partly due to the coupling of wind-generated noise into the sound channel by either a shallowing sound channel or a down-slope conversion process due to basin boundaries and sea mounts. Omnidirectional measurements made below the sound channel critical depth, in sparsely shipped basins, and at high sea states indicate two distinct regions divided by the occurrence of breaking waves. Prior to wave breaking, a possible sound generation mechanism is the interaction of surface waves and turbulence in the near-surface layer [I. Z. V. Gonchavov, *Atmos. Oceanic Phys.* 6(11) (1970); Yen and Perrone NUSC TR5833 (1979)]. Wave breaking produces bubble clouds. The dynamic evolution of these bubble clouds is a mechanism for the production of sound of frequencies less than 500 Hz. These clouds of micron-size bubbles are regions of low sonic velocity described by Wood's volume fraction equations. These regions can be treated as a compressible body with a composite mixture speed and density that can exhibit a collective resonant oscillation and radiate as monopole and dipole sources. However, due to the proximity of the sea surface, only the monopole and its image, an effective dipole, would be of importance. When driven by the wave breaking vorticity and turbulence, these regions are shown to result in sufficient radiated sound to produce noise levels comparable to those observed and are also pronounced scatterers of low-frequency sound.

Contributed Papers

2:15

KKK3. On ambient noise generation by "soliton" surface waves. Robert H. Mellen and David Middleton (Kildare Corporation, 95 Trumbull Street, New London, CT 06320)

Wave-wave interaction is one of the more promising mechanisms for wind-generated ambient noise. However, levels calculated with the linear

gravity/capillary dispersion model for surface waves are found to be much too low above about 10 Hz. This discrepancy may rest in the wave model rather than the mechanism itself. Surface-wave measurements in flumes show no evidence of dispersion at higher wavenumbers. To account for dispersionless propagation, a nonlinear model in which the surface fine structure is treated as a random ensemble of solitonlike hydraulic bumps rather than dispersive waves has been proposed. The "soliton"

model appears to account for measured acoustic backscattering strengths and associated Doppler spectra at moderate sea states. Effects of the model on ambient noise prediction are examined.

2:30

KKK4. Effect of monomolecular films on the underlying ocean ambient noise field at low sea states. Jim Rohr (Naval Ocean Systems Center, Code 634, San Diego, CA 92152) and Garr E. Updegraff (University of

California, Marine Physical Laboratory of the Scripps Institution of Oceanography, San Diego, CA 92152)

In sea states, two–six monomolecular films on the ocean surface have been previously reported to dramatically reduce the ambient noise field beneath them. The mechanism was presumed to be associated with ocean whitecapping. A recent series of low sea state tests with films, in the absence of whitecapping, has found an even more pronounced ambient noise reduction if the ocean surface agitation is above a certain critical level. The mechanism through which the films reduce surface noise at low sea states is explored.

2:45–3:10

Panel Discussion

Moderator: Douglas Cato

DSTO, Maritimes Systems Division, Darlinghurst, Australia

3:10–3:15

Break

Invited Papers

3:15

KKK5. Monitoring oceanic precipitation using ambient sound—An assessment. Jeffrey A. Nystuen (Code 68Ny, Department of Oceanography, Naval Postgraduate School, Monterey, CA 93943)

Several years ago, the idea that rainfall could be measured by monitoring the unique underwater sound generated by the rain was introduced. Two mechanisms by which raindrops can produce sound underwater have been identified. Data will be presented from current laboratory experiments exploring the relative importance of these two mechanisms for different sized drops at terminal fall velocities. The influence of wind on these mechanisms is not understood and currently prevents quantification of rainfall rate from ambient sound; however, oceanic data from the Canadian Atlantic Storms Program will be presented that demonstrate that precipitation can be detected acoustically even in conditions of high winds (20 m/s) and large waves (4–6 m).

3:35

KKK6. Underwater noise due to precipitation. Lawrence A. Crum, Hugh C. Pumphrey (National Center for Physical Acoustics, University of Mississippi, University, MS 38677), Andrea Prosperetti (Department of Mechanical Engineering, Johns Hopkins University, Baltimore, MD 21218), and Leif Bjorno (Industrial Acoustics Laboratory, Technical University of Denmark, DK-2800 Lyngby, Denmark)

In 1959, G. Franz published a thorough investigation of the underwater sound produced by liquid drop impacts [G. Franz, *J. Acoust. Soc. Am.* **31**, 1080 (1959)]. He discovered that, under certain conditions, a gas bubble was entrained by the impacting droplet, and the subsequent oscillation of this bubble resulted in a large amount of radiated sound. Recently, Scrimger has measured the underwater sound produced by rainfall and has discovered that a well-defined spectral peak exists near 15 kHz [J. A. Scrimger, *Nature* **318**, 647 (1985)]. The sound produced by the impact of water droplets on a water surface, both for individual and for multiple events such as those produced by artificial and natural rainfall, has been examined. The studies indicate that the major contribution to the underwater noise produced by both rain and snow is that associated with the oscillations of gas bubbles introduced into the water by the impact. Both experimental and theoretical evidence for these conclusions will be presented, including numerical studies of the drop impact process. [Work supported by the ONR.]

KKK7. Bubble clouds as sources and scatterers of underwater sound. A. Prosperetti, N. Q. Lu, and A. Lezzi (Department of Mechanical Engineering, The Johns Hopkins University, Baltimore, MD 21218)

A model of a bubble cloud based on a set of averaged equations is used to study the acoustic emission and scattering properties of these objects. From a study of the normal modes of the cloud, it is inferred that the lowest modes have a frequency much lower than that of the individual bubbles. This indicates that clouds of bubbles can radiate at such low frequencies if these modes are excited. The results of some scattering calculations at different wavelengths are also shown as a function of the incidence and scattering angles. [Work supported by ONR.]

Contributed Paper

4:15

KKK8. How raindrop impacts and air bubble resonances contribute to underwater sound spectra. Frédéric Laville (Groupe d'Acoustique de l'Université de Sherbrooke, Département de Génie Mécanique, Université de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada), Grayson D. Abbott, and Matthew J. Miller (Creare Incorporated, P. O. Box 71, Hanover, NH 03755)

Using underwater sound to measure the rate of rainfall over the oceans is a very promising approach. However, there are conflicting models on the spectral contributions of the two rainfall sound sources (raindrop

impacts on the water surface and air bubble resonances). For example, each of these two sources has been alternatively proposed as a major contributor to the spectrum as well as the origin of a spectral peak around 15 kHz. High-speed data acquisition and processing of underwater sound recorded in a lake under real rain and artificial raindrop conditions were used to identify the two sound sources in the time domain and determine their respective contributions to the long-term spectrum. Bubble resonances were found responsible for the spectral peak around 15 kHz and very dependent upon surface condition, whereas raindrop impacts were found responsible for a broadband spectrum rich in frequencies below 15 kHz and less dependent on surface conditions. [Work supported by DOE.]

4:30-5:00

Panel Discussion

Moderator: Lawrence A. Crum

National Center for Physical Acoustics, University, Mississippi

FRIDAY AFTERNOON, 26 MAY 1989

REGENCY B, 1:30 TO 4:00 P.M.

Session LLL. Late Arrivals

Junru Wu, Chairman

Department of Physics, University of Vermont, Burlington, Vermont 05405

Contributed Papers

1:30

LLL1. Streaming generated by aggregations in a rotating ultrasonic waveguide. Glenn Whitworth (Physics Department, University of Vermont, Burlington, VT 05405)

A chamber cavity, which has a square cross section and pressure-release walls, is used to produce a well-defined, 160-kHz standing ultrasonic field. A suspension of latex microspheres in aqueous metrizamide fills the chamber. The chamber rotates about a horizontal axis producing the centripetal force necessary to contain the buoyant spheres in the axial

region. At low particle concentrations, aggregations of microspheres form at half-wavelength intervals near the axial positions of acoustic pressure amplitude (PA) minima, as expected because of rotational and acoustic radiation forces. At higher concentrations, additional particle distributions are often seen that suggest the presence of flow. When high concentrations of larger particles are used, small aggregations also form at axial positions of PA maxima. To account for the flow, theory for acoustic streaming in a rotating fluid was developed and applied to flow near the larger aggregations. Reasonable agreement with observations was obtained when a term was added arising from drag on the aggregations. [Work supported by NIH Grant CA42947.]

LLL2. Experimental studies of a quasi two-dimensional nonpropagating hydrodynamic soliton. Erik M. Winkler and Junru Wu (Department of Physics, University of Vermont, Burlington, VT 05405)

A nonpropagating hydrodynamic soliton was first observed by Wu *et al.* [Phys. Rev. Lett. **52**, 1421 (1984)] in a rectangular water trough that was parametrically driven at a frequency below twice that of the linear cutoff frequency of (0,1) mode. A soliton of the same type was recently studied that forms when the trough is driven at a frequency below twice that of the linear cutoff frequency of (0,2) mode. The profile of the soliton in length direction as well as in width direction is determined. A video tape of the experiment will be shown during the presentation. [Work supported by NSF and Vermont Epscor.]

LLL3. Measurements of transmission loss to ultrasound through two-dimensional ensembles of trapped bubbles versus the amplitude of the incident wave. Junru Wu and Wesley L. Nyborg (Department of Physics, University of Vermont, Burlington, VT 05405)

In continuation of work reported previously, the frequency spectra of transmission coefficients to ultrasound through sheets of gas-filled micropores at incident amplitudes up to 3×10^4 Pa have been measured. It is found that as the amplitude of the incident wave is increased, the peak frequency of transmission loss through a bubble ensemble shifts to lower values by as much as 47%. Results of the experiments and the possible theoretical explanation will be presented and discussed. [Works of JW are supported by NSF and Vermont Epscor. Works of WLN are supported by NIH CA42947.]

LLL4. Acoustic radiation forces used to control the motion of a glass microneedle. Junru Wu (Department of Physics, University of Vermont, Burlington, VT 05405), Steven S. Work, and David M. Warshaw (Department of Physiology and Biophysics, University of Vermont, Burlington, VT 05405)

A method has been developed using a glass microneedle to tug on protein filaments and to measure their tensile strength (the force necessary to break the bond between individual protein molecules) [A. Kishino and T. Yanagida, Nature **334**, 74–76 (1988)]. To precisely control the motion of such a glass microneedle, a technique that uses acoustic radiation force at megahertz frequencies has been developed. Results from experiments using a rectangular resonant chamber and a video detection system have indicated that forces of 5 nN exerted on a glass microneedle and displacements of $4 \mu\text{m}$ can be resolved and were reproducible. A video tape of the experiment will be shown. [Works of JW supported by NSF and Vermont Epscor; DMW supported by NIH AM34872 and AHA.]

LLL5. Measured wind speed dependence of the underwater acoustic ambient vertical directional spectra at high frequencies. Robert M. Kennedy, Thomas K. Szlyk, and Susan M. Wentworth (Naval Underwater System Center, AUTEC, West Palm Beach, FL 33402-7517)

A measurement of the acoustic ambient arriving from a horizontal direction along with total acoustic intensity spectra allows one to infer both the total directional spectra and some physical characteristics of the sources of "sea surface sound." A long-term measurement of these two quantities was made at high frequency, i.e., 8 to 64 kHz, in the Tongue of

the Ocean, The Bahamas. The horizontally directed ambient was measured using vertically oriented line arrays and was observed for wind speeds ranging from 1 to 30 kn. The resulting database was used to estimate the statistics of anisotropic "noise gain" relative to the isotropic noise gain. The resulting normalized array performance was found to be a continuous function of total acoustic intensity and a discontinuous function of wind speed. Differences in the functional dependence and residual statistics were found for two cases: whitecaps present or not present. The relation of these results to the total directional spectra and a model of the near-surface distribution of acoustic sources are discussed.

LLL6. Modeling word-boundary phonological processes in continuous speech recognition. Briony Williams (Centre for Speech Technology Research, 80 South Bridge, Edinburgh EH1 1HN, United Kingdom) and Henry Thompson (Department of AI, 80 South Bridge, Edinburgh EH1 1HN, United Kingdom)

The automatic recognition of continuous English speech, as against isolated words, must contend with phonological processes at word boundaries that can distort the mapping between surface form and lexical entry. Past recognition systems have used generative phonological rules in a precompilation stage to expand the working dictionary. However, this greatly increases the bulk and complexity of the lexicon. It is also possible to use analytic rules to undo the putative effect of phonological processes at the time of recognition. However, this can lead to the postulation of nonwords and to slower processing times. A solution that blends the advantages of both approaches, with the disadvantages of neither, is to use finite state transducers (FSTs) as filters on permitted matchings between input strings and lexical entries. These were implemented in a recognition system, and found to increase the percentage of words recognized by three percentage points, at a cost of halving the signal-to-noise ratio, compared to the same system without phonological FSTs.

LLL7. Spontaneous emission of acoustic waves from shock fronts. F. Xu (Institute of Mechanics, Academia Sinica, Zhongguancun, Beijing 100080, People's Republic of China)

The criterion for spontaneous emission of acoustic waves from shock fronts,

$$\frac{1 - M^2 - (v_0/v_1)M^2}{1 - M^2 + (v_0/v_1)M^2} < \beta^2 \frac{dv}{dp} < 1 + 2M,$$

was obtained by D'yakov *et al.* [Zh. Eksp. Teor. Fiz. **27**, 288 (1954); V. M. Kontorovich, *ibid.* **33**, 1525 (1957); G. R. Fowles, Phys. Fluids **24**, 220 (1981)]. Based on the previous work of the interaction of acoustic waves with a shock front [F. Xu, Acta Mech. Sinica (English edition) **3**, 113 (1987)], a new criterion is obtained in this paper: $C_0 > \omega/k_y < C_1$. Then, the causes for the discrepancy between these two criteria are discussed and some comments on the previous papers are made. In addition, the amplification of the shock front is investigated and the results are compared with those of previous works [G. R. Fowles, Phys. Fluids **24**, 220 (1981); P. Harris, Tech. Rep. ARLCD-TR-78041 (1978)].

LLL8. Attenuation of Lamb waves in a plate loaded by a liquid under pressure. Paolo Diodati and Mete Severcan (Universita degli Studi di Perugia, Dipartimento di Fisica, Via Elce di Sotto, 06100 Perugia, Italy)

Attenuation of Lamb waves in a steel plate loaded on one side by a fluid was measured as a function of the fluid pressure. A monobloc, rec-

tangular, steel cavity was constructed and a 1-cm-thick steel plate was used to shut pressureproof the cavity which contained castor oil, $C_{11}H_{10}O_6$, in it. Acoustic pulses at 1.7 MHz were launched on the plate using a plastic wedge and a second wedge was used to detect the received pulses. The attenuation for different Lamb modes was observed to increase with pressure for all the modes except for the low-order modes, s_0 , a_0 , s_1 , a_1 , and a_2 , for which a decrease in attenuation was observed. Over the range of pressure used, considering the change in the wave velocities in the liquid and the steel plate, normally an increase in attenuation would be expected. The unexpected decrease in attenuation for low-order modes can be explained by the decrease in the power lost in the liquid near the plate boundary by the leaky waves, due to the shifting away from the phase-matching condition with the change in pressure.

3:30

LLL9. Acoustic emission from the Gran Sasso Mountain (AEGS). P. Diodati (Dipartimento di Fisica, Università di Perugia, Perugia, Italy), G. Paparo (Istituto di Acustica OM Corbino, Roma, Italy), and R. Scarpa (Dipartimento di Fisica, Università de l'Aquila, Aquila, Italy)

Rectangular parallelepiped samples extracted from a tunnel 10.2 km long and 1000 m above sea level and from the top of the highest mountain in the Appennines underwent laboratory studies for their acoustic emission (AE). The different responses are reported and explained. These are the preliminary results of the AEGS project which will involve collecting data from sensors located in different places of the mountain. In the center of the tunnel, where a laboratory of the National Institute of Nuclear

Physics (INFN) exists, was placed a seismometer (to tape record seismic events with a frequency up to 100 Hz), a geophone 3d (for oscillations up to 30 kHz), and four transducers (broadband, 50 kHz–2 MHz), for AE measurements. In a perforation made on the top and 1.5 km deep, at 1000 m from the surface, a seismometer, a geophone, and two transducers for AE were placed. In addition, on the top a geophone and two transducers for AE were placed. This acquisition system of seismic data will make it possible, by means of a computer network, to send all the data to a unique integrated system to analyze the physical parameters.

3:45

LLL10. Depth dependence of high-frequency ambient ocean noise. Albert B. Caron (Naval Underwater Systems Center, Newport, RI 02841)

Little or no published data exist on the depth dependence of ambient ocean noise at frequencies above 9 kHz. In this higher frequency region, absorption becomes a major effect in reducing surface-generated noise as receiver depth increases. To investigate this effect, two experiments were conducted. The first, conducted in the Tongue of the Ocean, off Andros Island in the Bahamas, utilized bottom-mounted hydrophones at a depth of 1500 m to observe ambient noise levels covering sea states 1–3 over a frequency span of 13 to 37 kHz. Near surface noise measurements were also made for comparison. The second experiment, conducted in Hawaii, covered a 9- to 25-kHz span with bottom-mounted hydrophones at depths of 751, 1173, 1769, 2481, 3981, and 4505 m. Results in both cases show a reduction in ambient noise levels with increasing depth and frequency, as predicted from models developed by Urlick and others. Closest agreement was observed with the assumption of a dipole surface intensity radiation pattern at the higher sea states.